IN THE UNITED STATES DISTRICT COURT
FOR THE EASTERN DISTRICT OF TEXAS
SHERMAN DIVISION

FAR NORTH PATENTS, LLC, Plaintiff, v. RIBBON COMMUNICATIONS INC. and SONUS NETWORKS, INC. d/b/a RIBBON COMMUNICATIONS OPERATING COMPANY, Defendants.	CIVIL ACTION NO. 4:19-cv-945 ORIGINAL COMPLAINT FOR PATENT INFRINGEMENT **JURY TRIAL DEMANDED**

ORIGINAL COMPLAINT FOR PATENT INFRINGEMENT

Plaintiff Far North Patents, LLC ("Far North Patents" or "Plaintiff") files this original complaint against Defendants Ribbon Communications Inc. and Sonus Networks, Inc. d/b/a Ribbon Communications Operating Company, (collectively, "Ribbon" or "Defendants"), alleging, based on its own knowledge as to itself and its own actions and based on information and belief as to all other matters, as follows:

PARTIES

1. Far North Patents is a limited liability company formed under the laws of the State of Texas, with its principal place of business at 18383 Preston Rd Suite 250, Dallas, Texas, 75252.

2. Defendant Ribbon Communications Inc. is a corporation organized and existing under the laws of Delaware. Ribbon Communications Inc. may be served through its registered agent, Corporation Service Company, at 251 Little Falls Drive, Wilmington, Delaware 19808.

3. Defendant Sonus Networks, Inc. d/b/a Ribbon Communications Operating Company is a corporation organized and existing under the laws of Delaware. Sonus Networks, Inc. d/b/a Ribbon Communications Operating Company may be served through its registered agent, Corporation Service Company, at 251 Little Falls Drive, Wilmington, Delaware 19808.

4. The Defendants identified in paragraphs 2-3 above (collectively, "Ribbon") are companies which together comprise one of the world's largest providers of real-time communications and networking services, software, and hardware.

5. The Ribbon defendants named above are part of the same corporate structure and distribution chain for the making, importing, offering to sell, selling, and/or using of the accused devices in the United States, including in the State of Texas generally and this judicial district in particular.

6. The Ribbon defendants named above share the same management, common ownership, advertising platforms, facilities, distribution chains and platforms, and accused product lines and products involving related technologies.

7. Thus, the Ribbon defendants named above operate as a unitary business venture and are jointly and severally liable for the acts of patent infringement alleged herein.

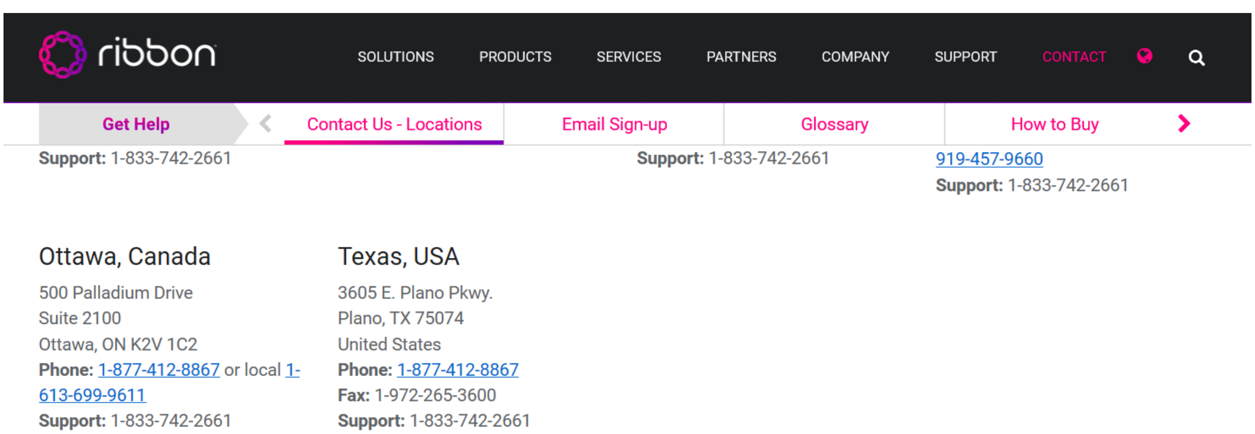
JURISDICTION AND VENUE

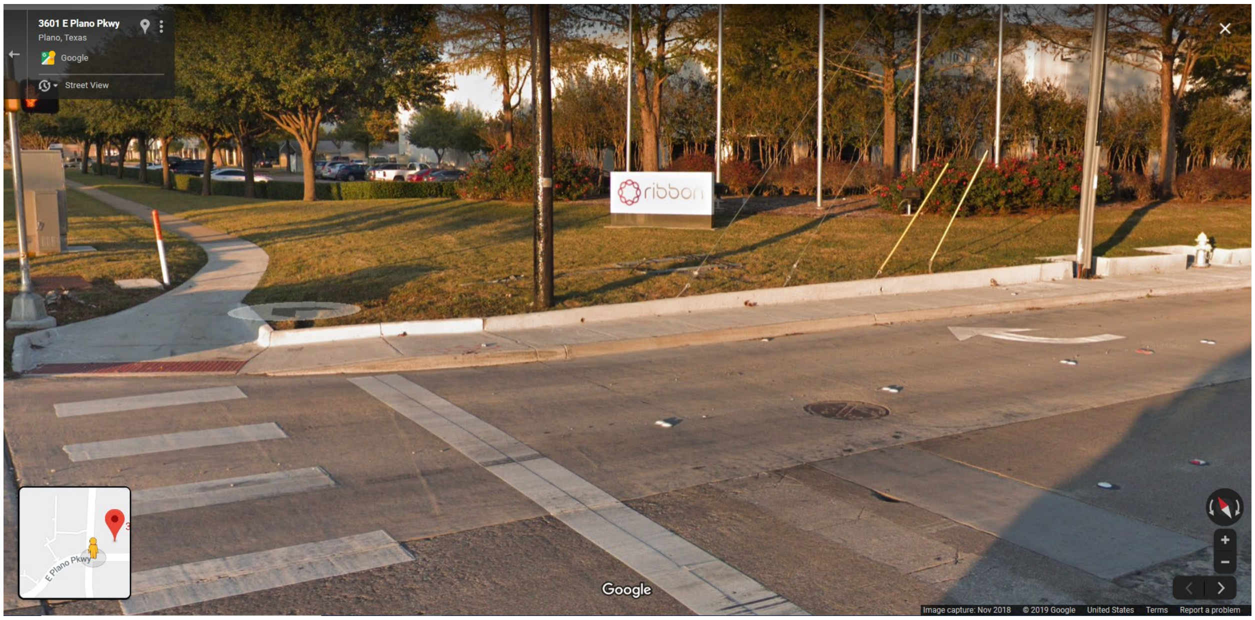
8. This is an action for infringement of United States patents arising under 35 U.S.C. §§ 271, 281, and 284–85, among others. This Court has subject matter jurisdiction of the action under 28 U.S.C. § 1331 and § 1338(a).

9. This Court has personal jurisdiction over Ribbon pursuant to due process and/or the Texas Long Arm Statute because, *inter alia*, (i) Ribbon has done and continues to do business in Texas; and (ii) Ribbon has committed and continues to commit acts of patent

infringement in the State of Texas, including making, using, offering to sell, and/or selling accused products in Texas, and/or importing accused products into Texas, including by Internet sales and sales via retail and wholesale stores, inducing others to commit acts of patent infringement in Texas, and/or committing a least a portion of any other infringements alleged herein.

10. Venue is proper in this district as to Ribbon Communications Inc. and Sonus Networks, Inc. d/b/a Ribbon Communications Operating Company pursuant to 28 U.S.C. § 1400(b). Venue is further proper because Ribbon has committed and continues to commit acts of patent infringement in this district, including making, using, offering to sell, and/or selling accused products in this district, and/or importing accused products into this district, including by Internet sales and sales via retail and wholesale stores, inducing others to commit acts of patent infringement in this district, and/or committing at least a portion of any other infringements alleged herein in this district. Ribbon also has a regular and established place of business in this district, including at 3605 E. Plano Pkwy., Plano, TX 75074 (as shown in the below screenshots from Ribbon's website, https://ribboncommunications.com/company/about-us/locations and from Google Maps Street View).





BACKGROUND

11. The patents-in-suit generally pertain to communications networks and other technology used in the provision of wireless services, Voice over Internet Protocol ("VoIP") phone systems, high speed networking, and other advanced communication services. The technology disclosed by the patents was developed by personnel at MCI WorldCom ("WorldCom"), Path1 Network Technologies Inc. ("Path1 Network Technologies"), Robelight LLC ("Robelight"), and BellSouth Corporation ("BellSouth").

12. WorldCom was a leading telecommunications service provider in the late 1990s and early 2000s. Verizon acquired WorldCom in 2005. The patents developed at WorldCom ("the Hardy patents") are related to Quality of Service ("QoS") evaluation in telecommunications systems.

13. The inventor of the Hardy patents, former principal analyst for quality measurement and analyses at WorldCom Dr. William C. Hardy, was at the forefront of QoS in telecommunications systems. Dr. Hardy developed, disclosed, and patented a solution for efficiently and consistently evaluating QoS. In fact, Dr. Hardy literally wrote the book on QoS

in telecommunications systems. *See* Hardy, William C., QoS Measurement and Evaluation of Telecommunications Quality of Service (Wiley 2001).

14. Dr. Hardy has received considerable praise for his work in QoS. Luis Sousa Cardoso, Quality of Service Development Group Chairman, left little doubt regarding the esteem with which he holds Dr. Hardy: "William C. 'Chris' Hardy is unquestionably among the leading lights in the field of QoS[.]" Dr. Hardy's book was reviewed in *IEEE Communications Magazine*, Vol. 40, No. 2, Feb. 2002, which stated that the book "provides a straightforward and very accessible approach to measurement and evaluation of QoS in telecommunications networks…strongly recommended for all people, either experiences professionals or graduates, involved in the area of networking[.]" He is even an honorary member of the Russian Academy of Science.

15. The Hardy patents (or the applications leading to them) have been cited during patent prosecution hundreds of times, by numerous leading companies in the computer networking and telecommunications industries industry, including Adtran, Alcatel-Lucent, Arris, AT&T, Avaya, Cisco, Deutsche Telekom (T-Mobile), Dolby Laboratories Licensing Corporation, Empirix, Ericsson, Genband, General Electric, IBM, Juniper, Microsoft, Motorola, NEC, Oracle, Panasonic, Ringcentral, Sharp, Siemens, Sprint, USAA, and Verizon.

16. Path1 Network Technologies is a provider of video over IP services and solutions. The patents developed at Path1 Network Technologies ("the Fellman patents") relate to providing service guarantees for time sensitive signals in computer networks. The inventors of these patents include Dr. Ronald D. Fellman and Dr. Rene L. Cruz. Drs. Fellman and Cruz, both former professors of electrical and computer engineering at the University of California at San Diego, were pioneers in network technology. Dr. Fellman was an IEEE Senior Member, and his

work was published in several IEEE Transactions journals, including IEEE Transactions on Networking, IEEE Transactions on Parallel and Distributed Systems, IEEE Transactions on Systems, Man, and Cybernetics, IEEE Transactions on Signal Processing, IEEE Transactions on Very Large Scale Integration (VLSI) Systems, IEEE Transactions on Acoustics, Speech and Signal Processing. He was also a co-founder of Path1 Network Technologies and of Qvidium Technologies. Dr. Cruz, a distinguished scholar in the field of communication networks, was said to have established the field of Network Calculus. In Dr. Cruz's election to be a Fellow of the IEEE in 2003, he was "cited for his expertise in the area of Quality-of-Service guarantees in packet-switched networks." http://jacobsschool.ucsd.edu/news/news_releases/release.sfe?id=1385.

17. The Fellman patents (or the applications leading to them) have been cited during patent prosecution hundreds of times, by numerous leading companies in the computer networking and telecommunications industries, including ABB Research, AMD, Amazon, AT&T, Atheros Communications, Avaya, Bose, Broadcom, Canon, Centurylink, Chi Mei Optoelectronics, Ciena, Cox Communications, Dell, F5 Networks, Fujitsu, Hitachi, Honeywell, Intel, IBM, Lucent, Lutron, Microsoft, National Instruments, National Semiconductor, NEC, Nortel Networks, Oceaneering, Phillips, Qualcomm, Robert Bosch, Samsung, Siemens, Sonos, Sony, Symantec, Texas Instruments, Toshiba, Ubiquiti Networks, Verizon, and Viasat.

18. The patents developed at Robelight ("the Light patents") relate to obtaining presence information over a network. Inventors Elliot D. Light and Jon L. Roberts are named inventors on over 30 patents combined. The Light patents (or the applications leading to them) have been cited during patent prosecution over a hundred times, by numerous leading companies in the computer networking and telecommunications industries, including Alcatel-Lucent, Apple,

AT&T, Avaya, Google, LG Electronics, Nortel Networks, Qualcomm, Rockstar Consortium, SAP, Shoretel, Vonage, and ZTE.

19. BellSouth, founded in 1983 as one of the seven original Regional Bell Operating Companies after the breakup of AT&T, was a giant in the telecommunications industry. BellSouth was active in both broadband and wireless, operating in the southern United States as well as in Argentina, Australia, Chile, Colombia, Ecuador, Guatemala, New Zealand, Nicaragua, Panama, Peru, Uruguay, and Venezuela. BellSouth was acquired by AT&T in 2006 for approximately $86 billion.

20. The patents developed at BellSouth ("the Easley patents") relate to providing a calling name service for mobile phones. Larry Scott Easley, the inventor of the Easley patents, was a prolific inventor for BellSouth—he was a named inventor on ten United States Patents. The Easley patents (or the applications leading to them) have been cited during patent prosecution over a hundred times, by numerous leading companies in the computer networking and telecommunications industries, including Alcatel-Lucent, AT&T, Ericsson, Genesys, Lucent, Nortel Networks, Siemens, Sprint, and Sybase 365.

COUNT I

DIRECT INFRINGEMENT OF U.S. PATENT NO. 8,689,105

21. On April 1, 2014, United States Patent No. 8,689,105 ("the '105 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Real-Time Monitoring of Perceived Quality of Packet Voice Transmission."

22. Far North Patents is the owner of the '105 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '105 Patent against infringers, and to collect damages for all relevant times.

23. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Sonus VX400 platform and Sonus/Ribbon SBC 1000/2000 session border controller families of products that include advanced quality monitoring capabilities (collectively, "accused products").

Introducing VX400

Overview

The VX400 platform supports analog, digital, and native IP voice, as well as port and trunk side serial data, making it a compelling solution for any government agency looking to deploy IP-based voice and data solutions. The product line combines the functionality of a media gateway, signaling control point, H.323/SIP inter-working device, media server and voice/data mux in a single chassis. As a port side interface, the optional serial data card enables legacy equipment to take advantage of the IP backbone, realizing full convergence.

Features

The VX400 uses dynamic flow control technology to overcome the challenges of maintaining secure VoIP calls in a degraded environment. The platform will keep secure calls connected even when the network experiences slow traffic flow and significant packet loss. Other implementations "relay" secure traffic as a normal (high bit rate) compressed audio call. Any lost, late or corrupt packets often result in the modem carrier slipping, causing the modem to retrain which typically results in a dropped call. The VX SERIES can withstand significant packet loss and jitter, and still keep the calls connected. FNBDT calls on VX400 can withstand complete network failure for up to 10 seconds without failing the call.

(Source : https://support.sonus.net/display/VXDOC/Introducing+VX400)

Basic RTCP Voice over IP Metrics

RTP control protocol (RTCP as per RFC 3550) measures the call quality and generates a report based on key metrics such as Packet loss rate/discard rate, Max*Jitter, Mean*Jitter, Average Round Trip Delay, Max Round Trip Delay, Burst length/density and gap length/density. RTCP is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol provides multiplexing of the data and control packets, for example using separate port numbers with UDP.

The primary function of RTCP involves providing feedback on the quality of the data distribution and sending reception feedback reports to all participants, which in turn will allow the QOS administrator to evaluate whether problems are local or global.

RTCP support on VX supports RFC 3550 and RFC 3611 with exception to the parameters listed under Sec 2.2 RFC Compliance. Following functionality is supported on VX:

- Building and Sending RTCP SR and RR report as per RFC 3550.
- Transmission interval for sending RTCP packets will be configurable.
- Receiving RTCP packets and extracting SR and RR report as per RFC 3550. VX will generate a CDR based report based on the metrics calculated
- Receiving RTCP*XR packet and extracting Statistical Summary and VoIP metrics as per RFC 3611.
- Sending of RTCP*XR packets will be supported with exceptions to the parameters listed under Sec 2.2 RFC Compliance.
- RTCP functionality will be supported for SIP transport only. For other protocols, VX will not process RTCP but will send and receive RTCP.
- RTCP metrics logs CDRs on per call basis.

(Source : https://support.sonus.net/display/VXDOC/Features+Added+in+VX+Release+4.3)

Sonus SBC 1000™ Session Border Controller

The award-winning SBC 1000 Session Border Controller delivers all of the functionality of the SBC 2000 in a solution right-sized for medium businesses and branch offices (up to 160 concurrent sessions). The SBC 1000 delivers built-in media transcoding, network security (encryption, authentication, DoS protection, etc.), robust SIP interworking, intelligent call routing, multi-vendor interoperability and 24/7/365 survivability. The Sonus SBC 1000 and the Sonus SBC 2000 Session Border Controllers are the only SBC solutions available which offer Microsoft Lync survivability through either a 3G/4G or PSTN connection to provide reliability in the event of a wide area network (WAN) failure.

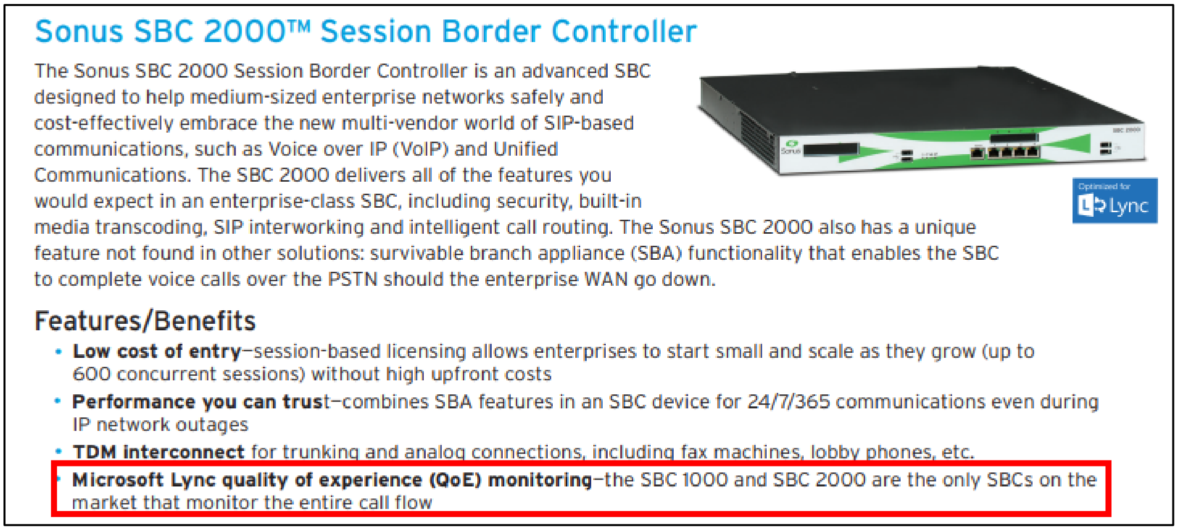
Optimized for Lync

Features/Benefits

- **Lowest cost of entry**–session-based licensing allows enterprises to get high-end SBC features at a low entry price
- **TDM interconnect** for trunking and analog connections, including fax machines, lobby phones, etc.
- **Get started quickly**–both the SBC 2000 and SBC 1000 can be provisioned and operational in as little as one hour
- **Microsoft Lync quality of experience (QoE) monitoring**–the SBC 1000 and SBC 2000 are the only SBCs on the market that monitor the entire call flow

(Source :

http://www.exertisgoconnect.nl/products/images/files/brochure_Sonus_SBC_Portfolio.pdf)



Sonus SBC 2000™ Session Border Controller

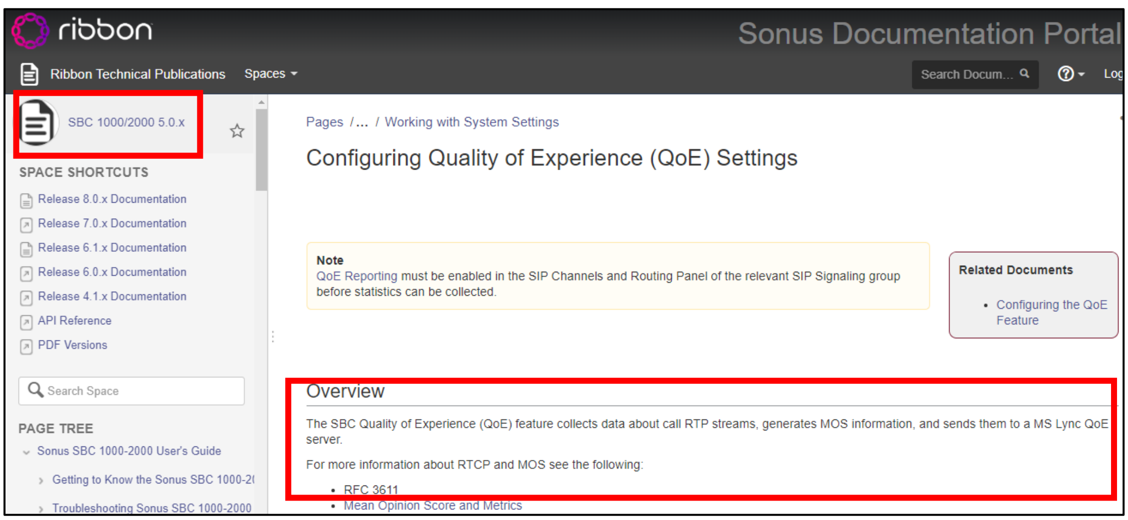
The Sonus SBC 2000 Session Border Controller is an advanced SBC designed to help medium-sized enterprise networks safely and cost-effectively embrace the new multi-vendor world of SIP-based communications, such as Voice over IP (VoIP) and Unified Communications. The SBC 2000 delivers all of the features you would expect in an enterprise-class SBC, including security, built-in media transcoding, SIP interworking and intelligent call routing. The Sonus SBC 2000 also has a unique feature not found in other solutions: survivable branch appliance (SBA) functionality that enables the SBC to complete voice calls over the PSTN should the enterprise WAN go down.

Features/Benefits

- **Low cost of entry**–session-based licensing allows enterprises to start small and scale as they grow (up to 600 concurrent sessions) without high upfront costs
- **Performance you can trust**–combines SBA features in an SBC device for 24/7/365 communications even during IP network outages
- **TDM interconnect** for trunking and analog connections, including fax machines, lobby phones, etc.
- **Microsoft Lync quality of experience (QoE) monitoring**–the SBC 1000 and SBC 2000 are the only SBCs on the market that monitor the entire call flow

(Source :

http://www.exertisgoconnect.nl/products/images/files/brochure_Sonus_SBC_Portfolio.pdf)



(Source :

https://support.sonus.net/display/UXDOC50/Configuring+Quality+of+Experience+%28QoE%29+Settings)

The Ribbon Communications SBC 1000™ Gateway

Protocol Support

- SNMPv2c, SNMPv3
- HTTPS
- SIP (RFC 3261) over UDP, TCP, TLS
- RTP/RTCP (RFC 3550, 3551)
- RTP/RTCP multiplexing over single UDP port (RFC 5761)
- DNS
- IPv4, IPv6, and IPv4/IPv6 interworking
- RIPv2, OSPF as dynamic IP routing protocols
- DHCP server
- DHCP client
- Asynchronous DNS for SIP
- NAT
- Support for Reason Header

(Source :

https://www.voipsupply.com/downloads/dl/file/id/37491/sbc_1000_gateway_datasheet.pdf)

The Ribbon Communications SBC 2000™

Protocol Support

- SNMPv2c, SNMPv3
- HTTPS
- RIPv2, OSPF as dynamic IP routing protocols
- SIP (RFC 3261) over UDP, TCP, TLS
- RTP/RTCP (RFC 3550, 3551)
- RTP/RTCP multiplexing over single UDP port (RFC 5761)
- IPv4, IPv6, and IPv4/IPv6 interworking
- DNS
- DHCP server
- DHCP client
- Asynchronous DNS for SIP
- NAT
- Support for Reason Header

(Source : https://www.voipsupply.com/downloads/dl/file/id/37471/sbc_2000_datasheet.pdf)

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RFC 3611                        RTCP XR                    November 2003

References

Normative References

   [1]  Bradner, S., "Key words for use in RFCs to Indicate Requirement
        Levels", BCP 14, RFC 2119, March 1997.

   [2]  Crocker, D., Ed. and P. Overell, "Augmented BNF for Syntax
        Specifications: ABNF", RFC 2234, November 1997.

   [3]  ETSI, "Quality of Service (QoS) measurement methodologies", ETSI
        TS 101 329-5 V1.1.1 (2000-11), November 2000.

   [4]  Handley, M. and V. Jacobson, "SDP: Session Description
        Protocol", RFC 2327, April 1998.

   [5]  Hovey, R. and S. Bradner, "The Organizations Involved in the
        IETF Standards Process", BCP 11, RFC 2028, October 1996.

   [6]  ITU-T, "The E-Model, a computational model for use in
        transmission planning", Recommendation G.107, January 2003.

   [7]  Narten, T. and H. Alvestrand, "Guidelines for Writing an IANA
        Considerations Section in RFCs", BCP 26, RFC 2434, October 1998.
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(Source : https://tools.ietf.org/html/rfc3611)

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  MOS-CQ: 8 bits
        The estimated mean opinion score for conversational quality
        (MOS-CQ) is defined as including the effects of delay and other
        effects that would affect conversational quality.  The metric
        may be calculated by converting an R factor determined
        according to ITU-T G.107 [6] or ETSI TS 101 329-5 [3] into an
        estimated MOS using the equation specified in G.107.  It is
        expressed as an integer in the range 10 to 50, corresponding to
        MOS x 10, as for MOS-LQ.

        A value of 127 indicates that this parameter is unavailable.
        Values other than 127 and the valid range defined above MUST
        not be sent and MUST be ignored by the receiving system.
```

(Source : https://tools.ietf.org/html/rfc3611)

24. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claims 1 and 23 of the ‘105 Patent. Ribbon’s infringement in this regard is ongoing.

25. Ribbon has infringed the ‘105 Patent by using the accused products and thereby practicing a method that includes obtaining, by a network device, a reference matrix based on

estimates of perceived audio quality of at least portions of one or more first packetized audio messages, the reference matrix modeling values of a plurality of characteristics associated with a particular quality level. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The quality of audio in VoIP networks (packet switched networks) are calculated using MOS (Mean Opinion score) values according to ITU-T G.107 Recommendation E-model. The E-model computes a transmission rating value R, which is a combinational effect of all the transmission parameters in an audio conversation. The E-model uses a reference table ("reference matrix") based on the estimates of perceived audio conversational/audio quality. The reference table includes modelling values like MOS-CQE (Mean Opinion Score – Estimated Conversational Quality), each associated with a quality level.

7 Target services

This Recommendation gives guidelines for QoE assessment of various telecommunication services mainly utilizing audio and visual media.

7.1 Audio

– Conversational voice and voice messaging

Speech communication services such as mobile telephony and voice over Internet protocol (VoIP), as well as conventional public switched telephone network (PSTN) and integrated services digital network (ISDN) services, are important targets of this Recommendation. The speech bandwidth can be either narrowband (NB) (300-3400 Hz) or wideband (WB) (100-7000 Hz).

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-G.1011-201506-S!!PDF-E&type=items)

8.5 Planning models

The input for planning models (Figure 8-5) includes the quality planning parameters of networks or terminals. It usually requires prior knowledge about the system under test. Such models can be applied to network planning and terminal/application design.

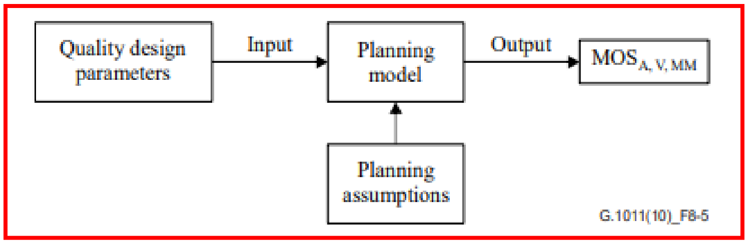

Figure 8-5 – Planning model

Standard examples of such models are [ITU-T G.107] for speech and [ITU-T G.1070] for videophone.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-G.1011-201506-S!!PDF-E&type=items)

Recommendation ITU-T G.107

The E-model: a computational model for use in transmission planning

1 Scope

This Recommendation describes a computational model, known as the E-model, that has proven useful as a transmission planning tool for assessing the combined effects of variations in several transmission parameters that affect conversational[1] quality of 3.1 kHz handset telephony. This computational model can be used, for example, by transmission planners to help ensure that users will be satisfied with end-to-end transmission performance whilst avoiding over-engineering of networks. It must be emphasized that the primary output from the model is the "rating factor" *R* but this can be transformed to give estimates of customer opinion. Such estimates are only made for transmission planning purposes and not for actual customer opinion prediction (for which there is no agreed-upon model recommended by the ITU-T).

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7 Structure and basic algorithms of the E-model

The E-model is based on the equipment impairment factor method, following previous transmission rating models. It was developed by an ETSI ad hoc group called "Voice Transmission Quality from Mouth to Ear".

The reference connection, as shown in Figure 1, is split into a send side and a receive side. The model estimates the conversational quality from mouth to ear as perceived by the user at the receive side, both as listener and talker.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

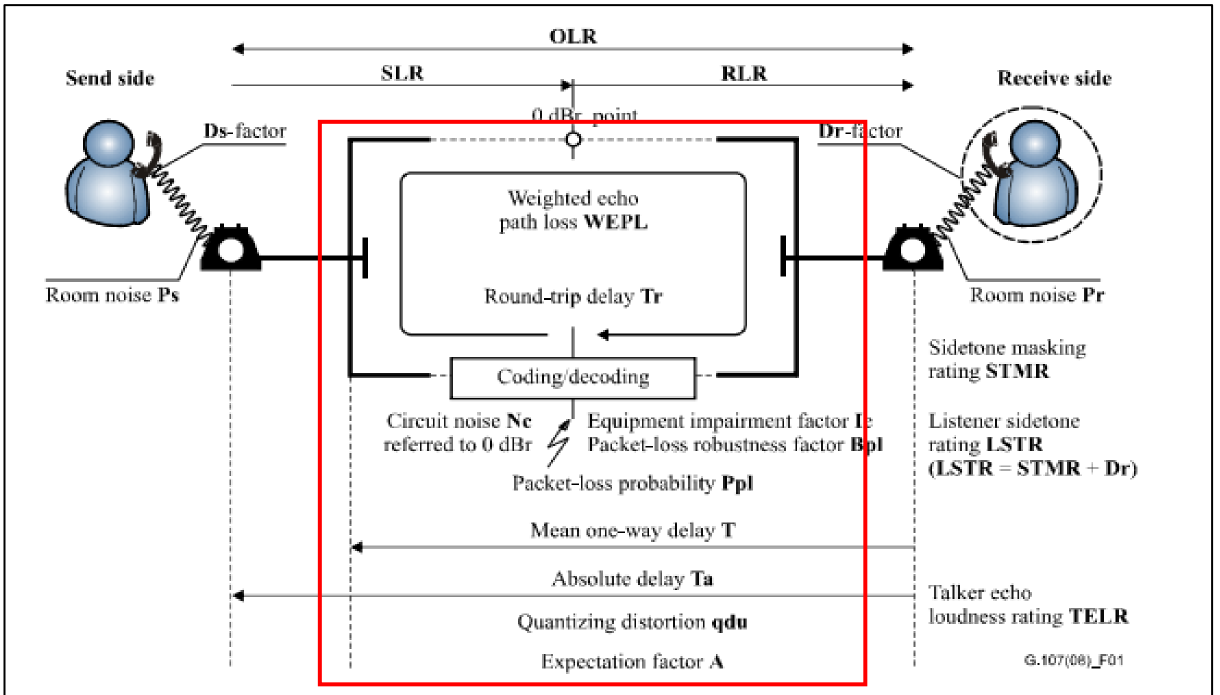

Figure 1 – Reference connection of the E-model

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model (see [b-ITU-T P-Sup.3]).

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the *R*-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

In some cases, transmission planners may not be familiar with the use of quality measures such as the *R* rating factor obtained from planning calculations, and thus provisional guidance for interpreting calculated *R* factors for planning purposes is given in Table B.1[3]. This table also contains equivalent transformed values of *R* into estimated conversational MOS_{CQE}, GoB and PoW.

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

***R*-value (lower limit)**	**MOS_{CQE} (lower limit)**	**GoB (%) (lower limit)**	**PoW (%) (upper limit)**	**User satisfaction**
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.2.3 MOS-CQE

The score is calculated by a network planning model which aims at predicting the quality in a conversational application situation. Estimates of conversational quality carried out according to [ITU-T G.107], when transformed to mean opinion score, give results in terms of MOS-CQE.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-P.800.1-201607-I!!PDF-E&type=items)

26. The methods practiced by Ribbon's use of the accused products include receiving, by the network device, one or more second packetized audio messages and evaluating, by the network device, at least portions of one or more of the one or more second packetized audio messages to obtain measurements associated with the plurality of characteristics. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The E-model is applied to a real-time voice call ("second packetized audio messages") for measuring its voice quality by calculating the R value. The R value can be converted into a MOS value. The R value represents the combinational effect of all transmission parameters in an audio conversation. The E-Model estimates the MOS-CQE/audio quality of the speech signals.

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model (see [b-ITU-T P-Sup.3]).

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

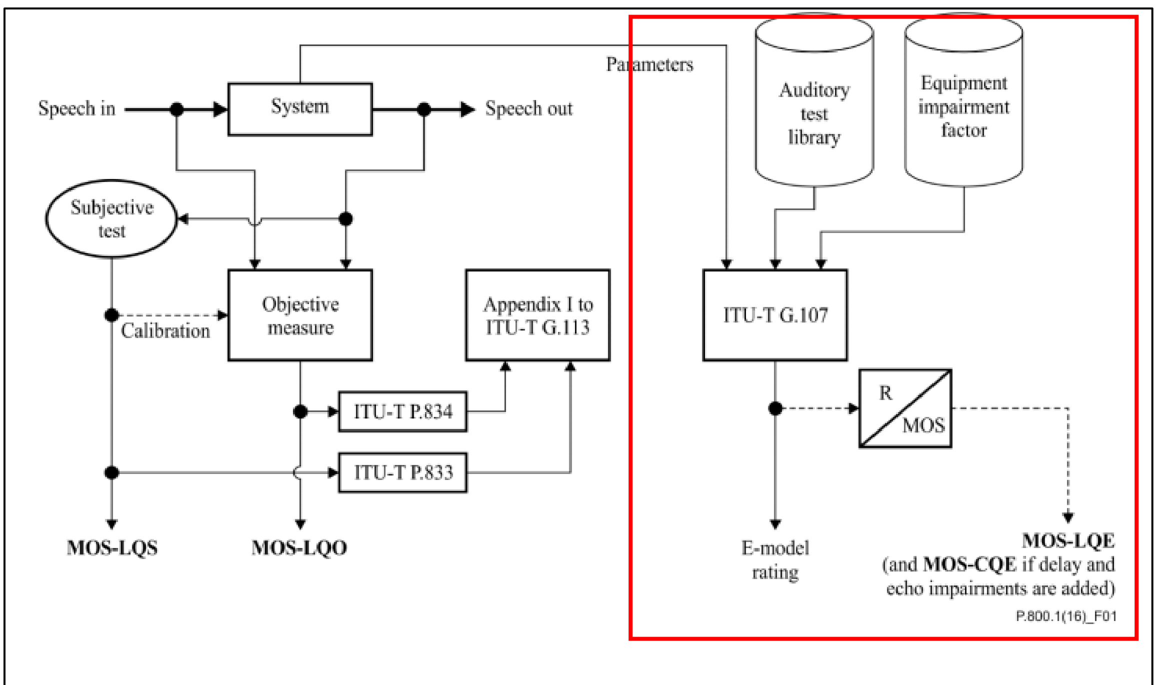

Figure 1 – Relationship between some audio MOS qualifiers

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-P.800.1-201607-I!!PDF-E&type=items)

27. The methods practiced by Ribbon's use of the accused products include creating, by the network device, a test matrix using the obtained measurements and comparing, by the network device, the test matrix and the reference matrix to predict a quality level associated with the one or more second packetized audio messages. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. ITU-T G.107 E-Model estimates MOS-CQE/audio quality of the speech signals. The test speech signal parameters are input to the G.107 E-Model for calculating the R and MOS values. The calculated R/MOS value ("test matrix") is then compared with the reference table ("reference matrix") for determining the perceived audio quality. For example, a comparison is performed between estimated MOS value and existing reference values to determine the perceived audio quality of the test speech. For instance, a MOS value of 4.5 and a R value of 95 is compared with each row of the reference table and a perceived voice quality is determined accordingly, which is Best/Very satisfied in this case.

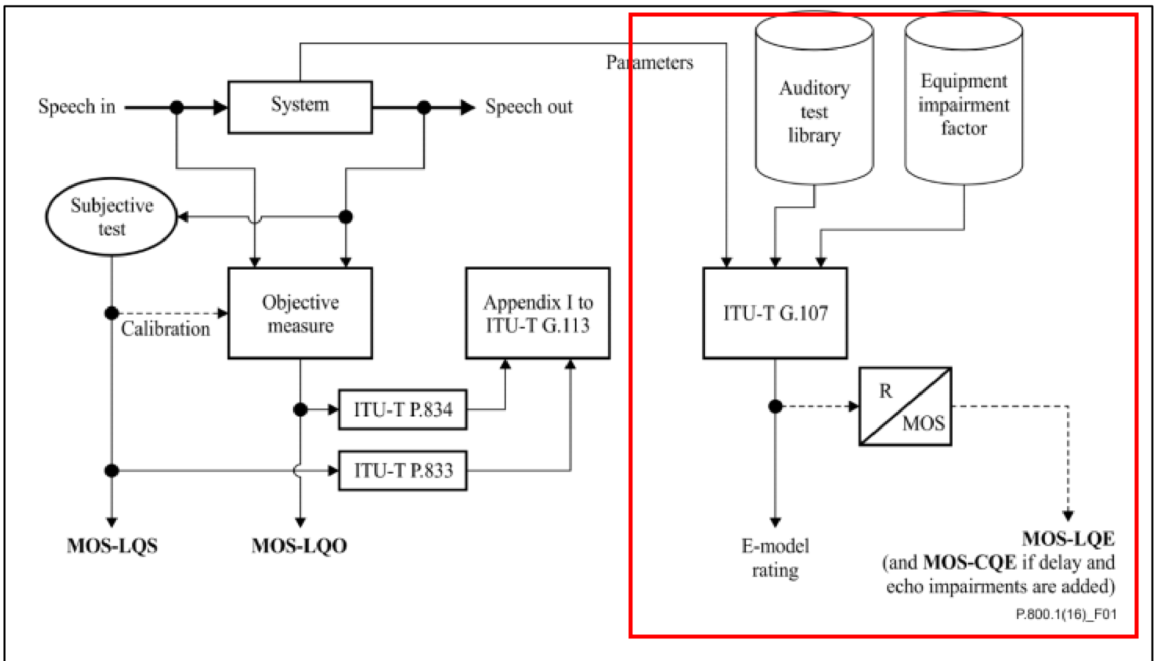

Figure 1 – Relationship between some audio MOS qualifiers

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-P.800.1-201607-I!!PDF-E&type=items)

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model (see [b-ITU-T P-Sup.3]).

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor R, which combines all transmission parameters relevant for the considered connection. This rating factor R is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

In some cases, transmission planners may not be familiar with the use of quality measures such as the R rating factor obtained from planning calculations, and thus provisional guidance for interpreting calculated R factors for planning purposes is given in Table B.1[3]. This table also contains equivalent transformed values of R into estimated conversational MOS_{CQE}, GoB and PoW.

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

R-value (lower limit)	MOS_{CQE} (lower limit)	GoB (%) (lower limit)	PoW (%) (upper limit)	User satisfaction
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

Table 1 - Definition of categories of speech transmission quality

Range of E-model Rating R	Speech transmission quality category	User satisfaction
90 ≤ R < 100	Best	Very satisfied
80 ≤ R < 90	High	Satisfied
70 ≤ R < 80	Medium	Some users dissatisfied
60 ≤ R < 70	Low	Many users dissatisfied
50 ≤ R < 60	Poor	Nearly all users dissatisfied

NOTE 1 – Connections with E-model Ratings R below 50 are not recommended.
NOTE 2 – Although the trend in transmission planning is to use E-model Ratings R, equations to convert E-model Ratings R into other metrics, e.g. %MOS, %GoB, PoW can be found in **ITU-T Rec. G.107 Annex B** [1].

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

28. Ribbon has infringed the '105 Patent by making, having made, using, importing, providing, supplying, distributing, selling or offering for sale products including the claimed non-transitory computer-readable medium having instructions stored thereon configured to cause a computing device to perform operations, and those operations including obtaining a reference matrix based on estimates of perceived audio quality of at least portions of one or more first packetized audio messages, the reference matrix modeling values of a plurality of characteristics associated with a particular quality level. For example, the accused products are configured to be used to implement the ITU-T G.107 Recommendation. The quality of audio in VoIP networks (packet switched networks) is calculated using MOS (Mean Opinion score) values according to ITU-T G.107 Recommendation E-model. The E-model computes a transmission rating value R, which is a combinational effect of all the transmission parameters in an audio conversation. The E-model uses a reference table ("reference matrix") based on the estimates of perceived audio conversational/audio quality. The reference table includes modelling values like MOS-CQE (Mean Opinion Score – Estimated Conversational Quality), each associated with a quality level.

7 Target services

This Recommendation gives guidelines for QoE assessment of various telecommunication services mainly utilizing audio and visual media.

7.1 Audio

– Conversational voice and voice messaging

Speech communication services such as mobile telephony and voice over Internet protocol (VoIP), as well as conventional public switched telephone network (PSTN) and integrated services digital network (ISDN) services, are important targets of this Recommendation. The speech bandwidth can be either narrowband (NB) (300-3400 Hz) or wideband (WB) (100-7000 Hz).

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-G.1011-201506-S!!PDF-E&type=items)

8.5 Planning models

The input for planning models (Figure 8-5) includes the quality planning parameters of networks or terminals. It usually requires prior knowledge about the system under test. Such models can be applied to network planning and terminal/application design.

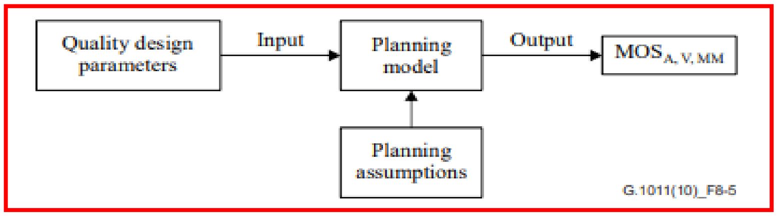

Figure 8-5 – Planning model

Standard examples of such models are [ITU-T G.107] for speech and [ITU-T G.1070] for videophone.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

Recommendation ITU-T G.107

The E-model: a computational model for use in transmission planning

1 Scope

This Recommendation describes a computational model, known as the E-model, that has proven useful as a transmission planning tool for assessing the combined effects of variations in several transmission parameters that affect conversational[1] quality of 3.1 kHz handset telephony. This computational model can be used, for example, by transmission planners to help ensure that users will be satisfied with end-to-end transmission performance whilst avoiding over-engineering of networks. It must be emphasized that the primary output from the model is the "rating factor" *R* but this can be transformed to give estimates of customer opinion. Such estimates are only made for transmission planning purposes and not for actual customer opinion prediction (for which there is no agreed-upon model recommended by the ITU-T).

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7 Structure and basic algorithms of the E-model

The E-model is based on the equipment impairment factor method, following previous transmission rating models. It was developed by an ETSI ad hoc group called "Voice Transmission Quality from Mouth to Ear".

The reference connection, as shown in Figure 1, is split into a send side and a receive side. The model estimates the conversational quality from mouth to ear as perceived by the user at the receive side, both as listener and talker.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

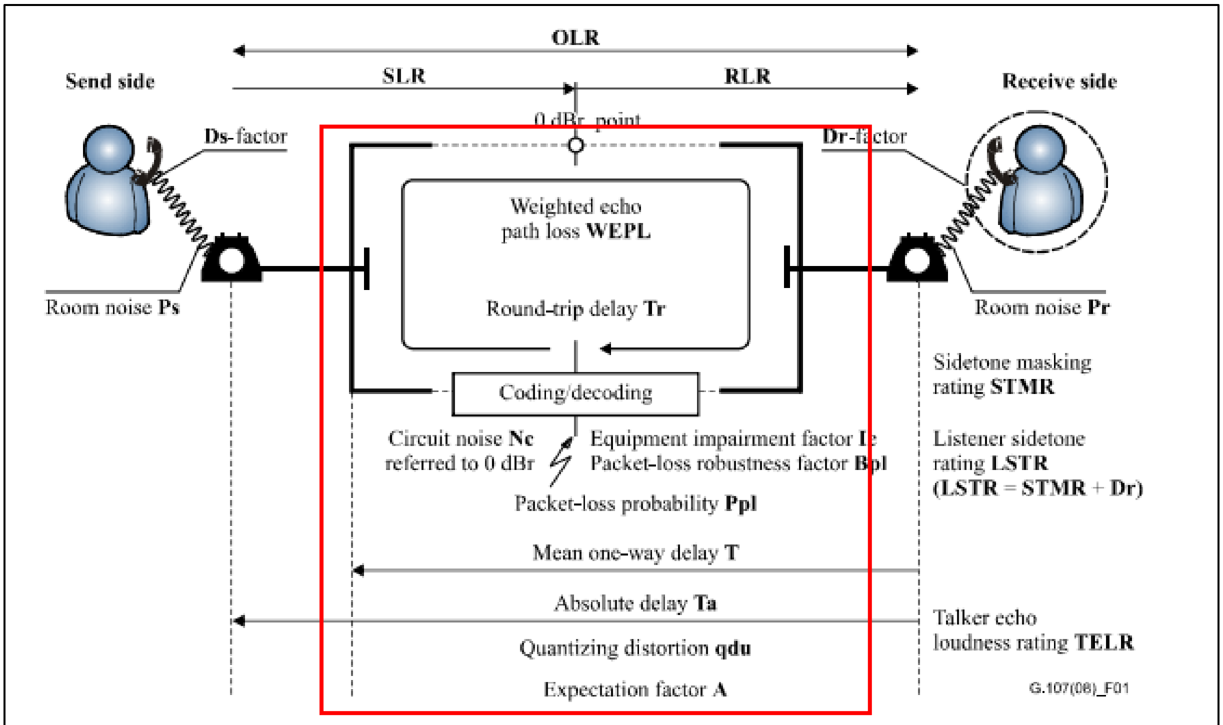

Figure 1 – Reference connection of the E-model

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model (see [b-ITU-T P-Sup.3]).

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \quad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the *R*-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

In some cases, transmission planners may not be familiar with the use of quality measures such as the *R* rating factor obtained from planning calculations, and thus provisional guidance for interpreting calculated *R* factors for planning purposes is given in Table B.1[3]. This table also contains equivalent transformed values of *R* into estimated conversational MOS_{CQE}, GoB and PoW.

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

***R*-value (lower limit)**	**MOS_{CQE} (lower limit)**	**GoB (%) (lower limit)**	**PoW (%) (upper limit)**	**User satisfaction**
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.2.3 MOS-CQE

The score is calculated by a network planning model which aims at predicting the quality in a conversational application situation. Estimates of conversational quality carried out according to [ITU-T G.107], when transformed to mean opinion score, give results in terms of MOS-CQE.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-P.800.1-201607-I!!PDF-E&type=items)

29. The operations performed by the accused products include creating a test matrix using measurements of at least portions of one or more second packetized audio messages associated with the plurality of characteristics and predicting a quality level associated with the

at least portions of one or more second packetized audio messages by comparing the test matrix to the reference matrix. For example, the accused products are configured to be used to implement the ITU-T G.107 Recommendation. The E-model is applied to a real-time voice call ("second packetized audio messages") for measuring its voice quality by calculating the R value. The R value can be converted into a MOS value. The R value represents the combinational effect of all transmission parameters in an audio conversation. ITU-T G.107 E-Model estimates MOS-CQE/audio quality of the speech signals. The test speech signal parameters are input to the G.107 E model for calculating the R and MOS values. The calculated R/MOS value ("test matrix") is then compared with the reference table ("reference matrix") for determining the perceived audio quality. For example, a comparison is performed between estimated MOS value and existing reference values to determine the perceived audio quality of the test speech. For instance, a MOS value of 4.5 and a R value of 95 would be compared with each row of the reference table and a perceived voice quality is determined accordingly, which is Best/Very satisfied in this case.

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model (see [b-ITU-T P-Sup.3]).

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the R-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

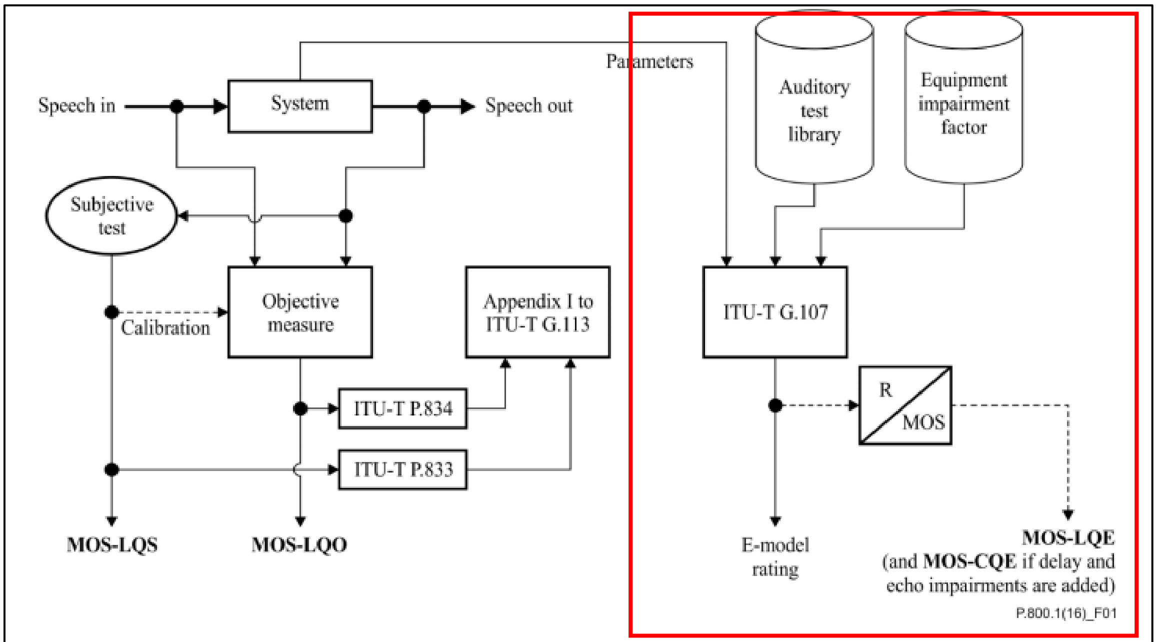

Figure 1 – Relationship between some audio MOS qualifiers

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-P.800.1-201607-I!!PDF-E&type=items)

In some cases, transmission planners may not be familiar with the use of quality measures such as the *R* rating factor obtained from planning calculations, and thus provisional guidance for interpreting calculated *R* factors for planning purposes is given in Table B.1[3]. This table also contains equivalent transformed values of *R* into estimated conversational MOS_{CQE}, GoB and PoW.

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

R-value (lower limit)	MOS_{CQE} (lower limit)	GoB (%) (lower limit)	PoW (%) (upper limit)	User satisfaction
90	4.34	97	~0	Very satisfied
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70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

Table 1 – Definition of categories of speech transmission quality

Range of E-model Rating R	Speech transmission quality category	User satisfaction
$90 \leq R < 100$	Best	Very satisfied
$80 \leq R < 90$	High	Satisfied
$70 \leq R < 80$	Medium	Some users dissatisfied
$60 \leq R < 70$	Low	Many users dissatisfied
$50 \leq R < 60$	Poor	Nearly all users dissatisfied
NOTE 1 – Connections with E-model Ratings R below 50 are not recommended. NOTE 2 – Although the trend in transmission planning is to use E-model Ratings R, equations to convert E-model Ratings R into other metrics, e.g. %MOS, %GoB, PoW can be found in **ITU-T Rec. G.107 Annex B** [1].		

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

30. Ribbon has had knowledge of the '105 Patent at least as of the date when it was notified of the filing of this action.

31. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

32. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the '105 Patent.

COUNT II

DIRECT INFRINGEMENT OF U.S. PATENT NO. 8,068,437

33. On November 29, 2011, United States Patent No. 8,068,437 ("the '437 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Determining the Effects of New Types of Impairments on Perceived Quality of a Voice Service."

34. Far North Patents is the owner of the '437 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '437 Patent against infringers, and to collect damages for all relevant times.

35. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Sonus VX400 platform and Sonus/Ribbon SBC 1000/2000 session border controller families of products that include advanced quality monitoring capabilities (collectively, "accused products").

Introducing VX400

Overview

The VX400 platform supports analog, digital, and native IP voice, as well as port and trunk side serial data, making it a compelling solution for any government agency looking to deploy IP-based voice and data solutions. The product line combines the functionality of a media gateway, signaling control point, H.323/SIP inter-working device, media server and voice/data mux in a single chassis. As a port side interface, the optional serial data card enables legacy equipment to take advantage of the IP backbone, realizing full convergence.

Features

The VX400 uses dynamic flow control technology to overcome the challenges of maintaining secure VoIP calls in a degraded environment. The platform will keep secure calls connected even when the network experiences slow traffic flow and significant packet loss. Other implementations "relay" secure traffic as a normal (high bit rate) compressed audio call. Any lost, late or corrupt packets often result in the modem carrier slipping, causing the modem to retrain which typically results in a dropped call. The VX SERIES can withstand significant packet loss and jitter, and still keep the calls connected. FNBDT calls on VX400 can withstand complete network failure for up to 10 seconds without failing the call.

(Source : https://support.sonus.net/display/VXDOC/Introducing+VX400)

Basic RTCP Voice over IP Metrics

RTP control protocol (RTCP as per RFC 3550) measures the call quality and generates a report based on key metrics such as Packet loss rate/discard rate, Max*Jitter, Mean*Jitter, Average Round Trip Delay, Max Round Trip Delay, Burst length/density and gap length/density. RTCP is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol provides multiplexing of the data and control packets, for example using separate port numbers with UDP.

The primary function of RTCP involves providing feedback on the quality of the data distribution and sending reception feedback reports to all participants, which in turn will allow the QOS administrator to evaluate whether problems are local or global.

RTCP support on VX supports RFC 3550 and RFC 3611 with exception to the parameters listed under Sec 2.2 RFC Compliance. Following functionality is supported on VX:

- Building and Sending RTCP SR and RR report as per RFC 3550.
- Transmission interval for sending RTCP packets will be configurable.
- Receiving RTCP packets and extracting SR and RR report as per RFC 3550. VX will generate a CDR based report based on the metrics calculated
- Receiving RTCP*XR packet and extracting Statistical Summary and VoIP metrics as per RFC 3611.
- Sending of RTCP*XR packets will be supported with exceptions to the parameters listed under Sec 2.2 RFC Compliance.
- RTCP functionality will be supported for SIP transport only. For other protocols, VX will not process RTCP but will send and receive RTCP.
- RTCP metrics logs CDRs on per call basis.

(Source : https://support.sonus.net/display/VXDOC/Features+Added+in+VX+Release+4.3)

Sonus SBC 1000™ Session Border Controller

The award-winning SBC 1000 Session Border Controller delivers all of the functionality of the SBC 2000 in a solution right-sized for medium businesses and branch offices (up to 160 concurrent sessions). The SBC 1000 delivers built-in media transcoding, network security (encryption, authentication, DoS protection, etc.), robust SIP interworking, intelligent call routing, multi-vendor interoperability and 24/7/365 survivability. The Sonus SBC 1000 and the Sonus SBC 2000 Session Border Controllers are the only SBC solutions available which offer Microsoft Lync survivability through either a 3G/4G or PSTN connection to provide reliability in the event of a wide area network (WAN) failure.

Features/Benefits

- **Lowest cost of entry**–session-based licensing allows enterprises to get high-end SBC features at a low entry price
- **TDM interconnect** for trunking and analog connections, including fax machines, lobby phones, etc.
- **Get started quickly**–both the SBC 2000 and SBC 1000 can be provisioned and operational in as little as one hour
- **Microsoft Lync quality of experience (QoE) monitoring**–the SBC 1000 and SBC 2000 are the only SBCs on the market that monitor the entire call flow

(Source :

http://www.exertisgoconnect.nl/products/images/files/brochure_Sonus_SBC_Portfolio.pdf)

Sonus SBC 2000™ Session Border Controller

The Sonus SBC 2000 Session Border Controller is an advanced SBC designed to help medium-sized enterprise networks safely and cost-effectively embrace the new multi-vendor world of SIP-based communications, such as Voice over IP (VoIP) and Unified Communications. The SBC 2000 delivers all of the features you would expect in an enterprise-class SBC, including security, built-in media transcoding, SIP interworking and intelligent call routing. The Sonus SBC 2000 also has a unique feature not found in other solutions: survivable branch appliance (SBA) functionality that enables the SBC to complete voice calls over the PSTN should the enterprise WAN go down.

Optimized for Lync

Features/Benefits

- **Low cost of entry**–session-based licensing allows enterprises to start small and scale as they grow (up to 600 concurrent sessions) without high upfront costs
- **Performance you can trust**–combines SBA features in an SBC device for 24/7/365 communications even during IP network outages
- **TDM interconnect** for trunking and analog connections, including fax machines, lobby phones, etc.
- **Microsoft Lync quality of experience (QoE) monitoring**–the SBC 1000 and SBC 2000 are the only SBCs on the market that monitor the entire call flow

(Source :

http://www.exertisgoconnect.nl/products/images/files/brochure_Sonus_SBC_Portfolio.pdf)

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SBC 1000/2000 5.0.x

SPACE SHORTCUTS

- Release 8.0.x Documentation
- Release 7.0.x Documentation
- Release 6.1.x Documentation
- Release 6.0.x Documentation
- Release 4.1.x Documentation
- API Reference
- PDF Versions

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- Sonus SBC 1000-2000 User's Guide
  - Getting to Know the Sonus SBC 1000-2(
  - Troubleshooting Sonus SBC 1000-2000

Pages /... / Working with System Settings

Configuring Quality of Experience (QoE) Settings

Note
QoE Reporting must be enabled in the SIP Channels and Routing Panel of the relevant SIP Signaling group before statistics can be collected.

Related Documents

- Configuring the QoE Feature

Overview

The SBC Quality of Experience (QoE) feature collects data about call RTP streams, generates MOS information, and sends them to a MS Lync QoE server.

For more information about RTCP and MOS see the following:

- RFC 3611
- Mean Opinion Score and Metrics

(Source :

https://support.sonus.net/display/UXDOC50/Configuring+Quality+of+Experience+%28QoE%29+Settings)

The Ribbon Communications SBC 1000™ Gateway

Protocol Support

- SNMPv2c, SNMPv3
- HTTPS
- SIP (RFC 3261) over UDP, TCP, TLS
- RTP/RTCP (RFC 3550, 3551)
- RTP/RTCP multiplexing over single UDP port (RFC 5761)
- DNS
- IPv4, IPv6, and IPv4/IPv6 interworking
- RIPv2, OSPF as dynamic IP routing protocols
- DHCP server
- DHCP client
- Asynchronous DNS for SIP
- NAT
- Support for Reason Header

(Source :

https://www.voipsupply.com/downloads/dl/file/id/37491/sbc_1000_gateway_datasheet.pdf)

The Ribbon Communications SBC 2000™

Protocol Support

- SNMPv2c, SNMPv3
- HTTPS
- RIPv2, OSPF as dynamic IP routing protocols
- SIP (RFC 3261) over UDP, TCP, TLS
- RTP/RTCP (RFC 3550, 3551)
- RTP/RTCP multiplexing over single UDP port (RFC 5761)
- IPv4, IPv6, and IPv4/IPv6 interworking
- DNS
- DHCP server
- DHCP client
- Asynchronous DNS for SIP
- NAT
- Support for Reason Header

(Source : https://www.voipsupply.com/downloads/dl/file/id/37471/sbc_2000_datasheet.pdf)

```
RFC 3611                        RTCP XR                    November 2003

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   [6]  ITU-T, "The E-Model, a computational model for use in
        transmission planning", Recommendation G.107, January 2003.

   [7]  Narten, T. and H. Alvestrand, "Guidelines for Writing an IANA
        Considerations Section in RFCs", BCP 26, RFC 2434, October 1998.
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(Source : https://tools.ietf.org/html/rfc3611)

```
   MOS-CQ: 8 bits
         The estimated mean opinion score for conversational quality
         (MOS-CQ) is defined as including the effects of delay and other
         effects that would affect conversational quality.  The metric
         may be calculated by converting an R factor determined
         according to ITU-T G.107 [6] or ETSI TS 101 329-5 [3] into an
         estimated MOS using the equation specified in G.107.  It is
         expressed as an integer in the range 10 to 50, corresponding to
         MOS x 10, as for MOS-LQ.

         A value of 127 indicates that this parameter is unavailable.
         Values other than 127 and the valid range defined above MUST
         not be sent and MUST be ignored by the receiving system.
```

(Source : https://tools.ietf.org/html/rfc3611)

36. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 9 of the '437 Patent. Ribbon's infringement in this regard is ongoing.

37. Ribbon has infringed the '437 Patent by using the accused products and thereby practicing a method performed by a computer system that includes generating, by a processor of the computer system, an assumed model for a second communication service, where the assumed

model is used to transform data regarding a first performance characteristic in the second communication service to reflect effects from a second performance characteristic in the second communication service. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The ITU-T G.107 Recommendation includes an E-model for calculating voice quality as perceived by a typical telephone user. The E-model outputs a transmission rating factor i.e., R, which can be transformed into Mean Opinion Score i.e., MOS value that represents the voice quality. The R value combines the effects of all relevant transmission parameters, and comprises of an effective Equipment impairment factor, $I_{e\text{-}eff}$. The E-model is applied to a real-time voice call ("second communication service") for measuring its voice quality. The effective Equipment impairment factor is calculated using a mathematical algorithm ("assumed model"). The mathematical algorithm includes an addition of two values. The first value is an equipment impairment factor ("first performance characteristic") at zero packet loss, or I_e. The I_e values are based on subjective MOS test results and are predefined for different codecs in ITU-T G.113 recommendation. The second value is a computation of different packet-loss-based parameters ("second performance characteristic") namely, a packet loss robustness factor (Bpl), packet loss probability (Ppl) and a burst ratio. Thus, the computed $I_{e\text{-}eff}$ value reflects the effects of packet loss in the voice quality.

The E-model: a computational model for use in transmission planning

1 Scope

This Recommendation describes a computational model, known as the E-model, that has proven useful as a transmission planning tool for assessing the combined effects of variations in several transmission parameters that affect the conversational[1] quality of 3.1 kHz handset telephony. This computational model can be used, for example, by transmission planners to help ensure that users will be satisfied with end-to-end transmission performance whilst avoiding over-engineering of networks. It must be emphasized that the primary output from the model is the "rating factor" *R* but this can be transformed to give estimates of customer opinion. Such estimates are only made for transmission planning purposes and not for actual customer opinion prediction (for which there is no agreed-upon model recommended by the ITU-T).

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the R-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

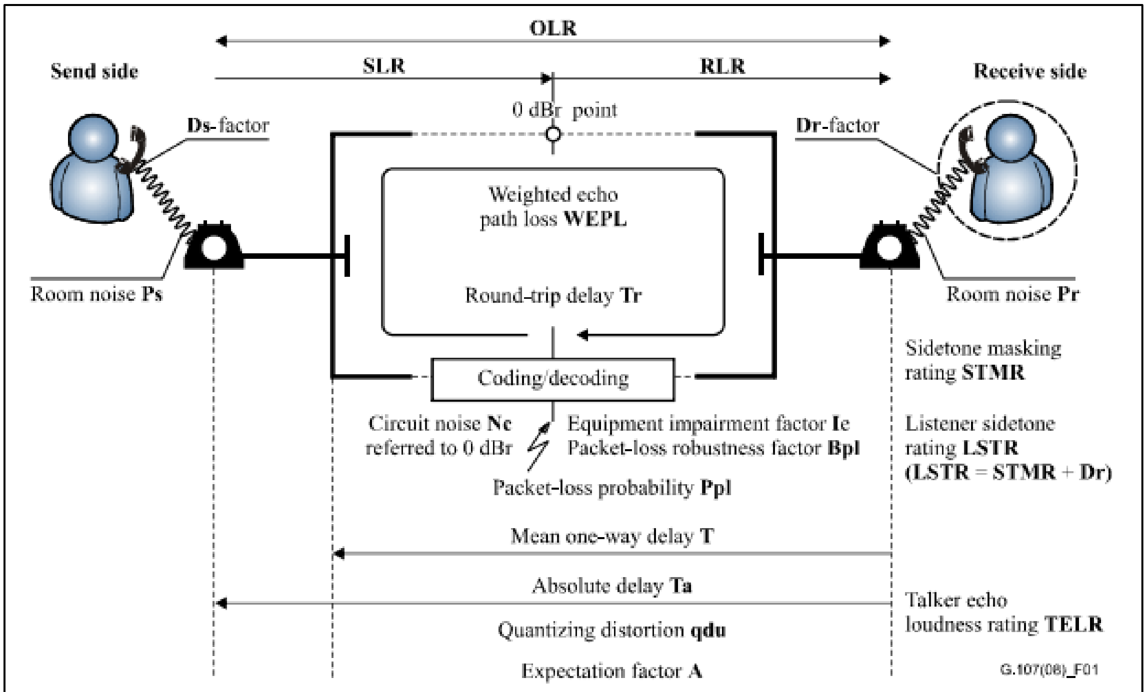

Figure 1 – Reference connection of the E-model

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \quad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.5 Equipment impairment factor, Ie

The values for the equipment impairment factor *Ie* of elements using low bit-rate codecs are not related to other input parameters. They depend on subjective mean opinion score (MOS) test results as well as on network experience. Refer to Appendix I of [ITU-T G.113] for the currently recommended values of *Ie*.

Specific impairment factor values for codec operation under random[2] packet-loss have formerly been treated using tabulated, packet-loss dependent *Ie*-values. Now, the packet-loss robustness factor *Bpl* is defined as a codec-specific value. The packet-loss dependent effective equipment impairment factor *Ie-eff* is derived using the codec-specific value for the equipment impairment factor at zero packet-loss *Ie* and the packet-loss robustness factor *Bpl*, both listed in Appendix I of [ITU-T G.113] for several codecs. With the packet-loss probability *Ppl*, *Ie-eff* is calculated using the equation:

$$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \quad (7\text{-}29)$$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

This appendix provides up-to-date information on available values of the equipment impairment factor, *Ie*, and packet-loss robustness factor, *Bpl*, for codecs or codec families. It is intended to be updated regularly.

Table I.1 provides provisional planning values for the equipment impairment factor, *Ie*. These *Ie* values refer to non-error conditions without propagation errors, frame-erasures or packet loss. Subsequent tables deal with error and various loss conditions.

Table I.1 – Provisional planning values for the equipment impairment factor, *Ie*

Codec type	Reference	Operating rate [kbit/s]	*Ie* value
PCM (see Note)	G.711	64	0
ADPCM	G.726, G.727	40	2
	G.721, G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-G.113-200711-I!!PDF-E&type=items)

38. The methods practiced by Ribbon's use of the accused products include establishing, by the processor, a communication session via the second communication service and obtaining, by the processor, subjective ratings of the first performance characteristic in the second communication service using the established communication session. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The E-model is applied to a real-time voice call session over a system including the accused products ("second communication service") for measuring the call's voice quality by calculating the R value. The R value comprises of an effective Equipment impairment factor, $I_{e\text{-}eff}$ which is calculated using various parameters like an equipment impairment factor at zero packet loss I_e ("first performance characteristic"), and other packet loss based parameters. The I_e values ("subjective ratings") are derived from the results of subjective listening-only tests and are used

as an input to the E-Model. They can be obtained from predefined values based on the implemented codec.

> **7.1 Calculation of the transmission rating factor, R**
>
> According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].
>
> Psychological factors on the psychological scale are additive.
>
> The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:
>
> $$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$
>
> *Ro* represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

> **7.5 Equipment impairment factor, Ie**
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>
> Specific impairment factor values for codec operation under random[2] packet-loss have formerly been treated using tabulated, packet-loss dependent *Ie*-values. Now, the packet-loss robustness factor *Bpl* is defined as a codec-specific value. The packet-loss dependent effective equipment impairment factor *Ie-eff* is derived using the codec-specific value for the equipment impairment factor at zero packet-loss *Ie* and the packet-loss robustness factor *Bpl*, both listed in Appendix I of [ITU-T G.113] for several codecs. With the packet-loss probability *Ppl*, *Ie-eff* is calculated using the equation:
>
> $$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{\dfrac{Ppl}{BurstR} + Bpl} \qquad (7\text{-}29)$$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

This appendix provides up-to-date information on available values of the equipment impairment factor, *Ie*, and packet-loss robustness factor, *Bpl*, for codecs or codec families. It is intended to be updated regularly.

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	G.721, G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-G.113-200711-I!!PDF-E&type=items)

39. The methods practiced by Ribbon's use of the accused products include generating, by the processor, altered subjective ratings using the assumed model to reflect effects of the second performance characteristic on the subjective ratings. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The effective equipment impairment factor $I_{e\text{-}eff}$ is calculated using a mathematical algorithm ("assumed model"). The mathematical algorithm includes an addition of two values. The first value is an equipment impairment factor ("first performance characteristic") at zero packet loss i.e., I_e. The I_e values are based on subjective MOS test results and are predefined for different codecs in ITU-T G.113 recommendation. The second value is a computation of different packet loss ("second performance characteristic") based parameters namely, a packet loss robustness factor (Bpl), packet loss probability (Ppl) and a burst ratio. Thus, the computed $I_{e\text{-}eff}$ value reflects the effects of packet loss on the equipment impairment factor at zero packet loss.

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \tag{7-1}$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.5 Equipment impairment factor, Ie

The values for the equipment impairment factor *Ie* of elements using low bit-rate codecs are not related to other input parameters. They depend on subjective mean opinion score (MOS) test results as well as on network experience. Refer to Appendix I of [ITU-T G.113] for the currently recommended values of *Ie*.

Specific impairment factor values for codec operation under random[2] packet-loss have formerly been treated using tabulated, packet-loss dependent *Ie*-values. Now, the packet-loss robustness factor *Bpl* is defined as a codec-specific value. The packet-loss dependent effective equipment impairment factor *Ie-eff* is derived using the codec-specific value for the equipment impairment factor at zero packet-loss *Ie* and the packet-loss robustness factor *Bpl*, both listed in Appendix I of [ITU-T G.113] for several codecs. With the packet-loss probability *Ppl*, *Ie-eff* is calculated using the equation:

$$Ie\text{-}eff = Ie + (95 - Ie) \cdot \frac{Ppl}{\frac{Ppl}{BurstR} + Bpl} \tag{7-29}$$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

This appendix provides up-to-date information on available values of the equipment impairment factor, *Ie*, and packet-loss robustness factor, *Bpl*, for codecs or codec families. It is intended to be updated regularly.

Table I.1 provides provisional planning values for the equipment impairment factor, *Ie*. These *Ie* values refer to non-error conditions without propagation errors, frame-erasures or packet loss. Subsequent tables deal with error and various loss conditions.

Table I.1 – Provisional planning values for the equipment impairment factor, *Ie*

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ADPCM	G.726, G.727	40	2
	G.721, G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-G.113-200711-I!!PDF-E&type=items)

40. The methods practiced by Ribbon's use of the accused products include generating, by the processor, quality index values from the altered subjective ratings. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The MOS_{CQE} values are calculated using the R values. The R value is calculated using various parameters which includes the effective equipment impairment factor $I_{e\text{-}eff}$ ("altered subjective rating").

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the *R*-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

41. The methods practiced by Ribbon's use of the accused products include comparing, by the processor, the generated quality index values to quality index values of a first communication service and determining, by the processor, whether the quality of the second communication service is comparable to a quality of the first communication service based on the comparison. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The E-model is based on modeling the results from multiple subjective

tests performed on a wide range of transmission parameters. It also includes a reference table with different R value and MOS value thresholds, and corresponding perceived voice quality. The MOS values (quality index) in the reference table are obtained using an aggregate of multiple test calls' ("first communication service") data. The computed MOS value is then compared with the reference table. Based on the comparison, it is determined whether the computed MOS value is comparable to the reference MOS value—e.g., whether the second communication service is expected to fall into the same user satisfaction category as the first communication service.

The E-model (**ITU-T Rec. G.107** [1]) is a transmission planning tool that provides a prediction of the expected voice quality, as perceived by a typical telephone user, for a complete end-to-end (i.e. mouth-to-ear) telephone connection under conversational conditions. The E-model takes into account a wide range of telephony-band impairments, in particular the impairment due to low bit-rate coding devices and one-way delay, as well as the "classical" telephony impairments of loss, noise and echo. It can be applied to assess the voice quality of wireline and wireless scenarios, based on circuit-switched and packet-switched technology.

The E-model is based on modeling the results from a large number of subjective tests done in the past on a wide range of transmission parameters. The primary output of the E-model calculations is a scalar quality rating value known as the "Transmission Rating Factor, R". R can be transformed into other quality measures such as Mean Opinion Score (MOS-CQE [2]), Percentage Good or Better ((GoB) or Percentage Poor or Worse ((PoW). However, caution should be exercised when comparing these transformed measures with values of MOS, %GoB or %PoW from other sources, which may not have been obtained under comparable conditions.

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

R-value (lower limit)	MOS_{CQE} (lower limit)	GoB (%) (lower limit)	PoW (%) (upper limit)	User satisfaction
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the R-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

Table 1 - Definition of categories of speech transmission quality

Range of E-model Rating R	Speech transmission quality category	User satisfaction
90 ≤ R < 100	Best	Very satisfied
80 ≤ R < 90	High	Satisfied
70 ≤ R < 80	Medium	Some users dissatisfied
60 ≤ R < 70	Low	Many users dissatisfied
50 ≤ R < 60	Poor	Nearly all users dissatisfied

NOTE 1 – Connections with E-model Ratings R below 50 are not recommended.
NOTE 2 – Although the trend in transmission planning is to use E-model Ratings R, equations to convert E-model Ratings R into other metrics, e.g. %MOS, %GoB, PoW can be found in **ITU-T Rec. G.107 Annex B** [1].

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

42. Far North Patents only asserts method claims from the ‘437 Patent.

43. Ribbon has had knowledge of the ‘437 Patent at least as of the date when it was notified of the filing of this action.

44. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

45. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the ‘437 Patent.

COUNT III

DIRECT INFRINGEMENT OF U.S. PATENT NO. 7,085,230

46. On August 1, 2006, United States Patent No. 7,085,230 ("the '230 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Method and System for Evaluating the Quality of Packet-Switched Voice Signals."

47. Far North Patents is the owner of the '230 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '230 Patent against infringers, and to collect damages for all relevant times.

48. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Sonus VX400 platform and Sonus/Ribbon SBC 1000/2000 session border controller families of products that include advanced quality monitoring capabilities (collectively, "accused products").

Introducing VX400

Overview

The VX400 platform supports analog, digital, and native IP voice, as well as port and trunk side serial data, making it a compelling solution for any government agency looking to deploy IP-based voice and data solutions. The product line combines the functionality of a media gateway, signaling control point, H.323/SIP inter-working device, media server and voice/data mux in a single chassis. As a port side interface, the optional serial data card enables legacy equipment to take advantage of the IP backbone, realizing full convergence.

Features

The VX400 uses dynamic flow control technology to overcome the challenges of maintaining secure VoIP calls in a degraded environment. The platform will keep secure calls connected even when the network experiences slow traffic flow and significant packet loss. Other implementations "relay" secure traffic as a normal (high bit rate) compressed audio call. Any lost, late or corrupt packets often result in the modem carrier slipping, causing the modem to retrain which typically results in a dropped call. The VX SERIES can withstand significant packet loss and jitter, and still keep the calls connected. FNBDT calls on VX400 can withstand complete network failure for up to 10 seconds without failing the call.

(Source : https://support.sonus.net/display/VXDOC/Introducing+VX400)

Basic RTCP Voice over IP Metrics

RTP control protocol (RTCP as per RFC 3550) measures the call quality and generates a report based on key metrics such as Packet loss rate/discard rate, Max*Jitter, Mean*Jitter, Average Round Trip Delay, Max Round Trip Delay, Burst length/density and gap length/density. RTCP is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol provides multiplexing of the data and control packets, for example using separate port numbers with UDP.

The primary function of RTCP involves providing feedback on the quality of the data distribution and sending reception feedback reports to all participants, which in turn will allow the QOS administrator to evaluate whether problems are local or global.

RTCP support on VX supports RFC 3550 and RFC 3611 with exception to the parameters listed under Sec 2.2 RFC Compliance. Following functionality is supported on VX:

- Building and Sending RTCP SR and RR report as per RFC 3550.
- Transmission interval for sending RTCP packets will be configurable.
- Receiving RTCP packets and extracting SR and RR report as per RFC 3550. VX will generate a CDR based report based on the metrics calculated
- Receiving RTCP*XR packet and extracting Statistical Summary and VoIP metrics as per RFC 3611.
- Sending of RTCP*XR packets will be supported with exceptions to the parameters listed under Sec 2.2 RFC Compliance.
- RTCP functionality will be supported for SIP transport only. For other protocols, VX will not process RTCP but will send and receive RTCP.
- RTCP metrics logs CDRs on per call basis.

(Source : https://support.sonus.net/display/VXDOC/Features+Added+in+VX+Release+4.3)

Sonus SBC 1000™ Session Border Controller

The award-winning SBC 1000 Session Border Controller delivers all of the functionality of the SBC 2000 in a solution right-sized for medium businesses and branch offices (up to 160 concurrent sessions). The SBC 1000 delivers built-in media transcoding, network security (encryption, authentication, DoS protection, etc.), robust SIP interworking, intelligent call routing, multi-vendor interoperability and 24/7/365 survivability. The Sonus SBC 1000 and the Sonus SBC 2000 Session Border Controllers are the only SBC solutions available which offer Microsoft Lync survivability through either a 3G/4G or PSTN connection to provide reliability in the event of a wide area network (WAN) failure.

Features/Benefits

- **Lowest cost of entry**–session-based licensing allows enterprises to get high-end SBC features at a low entry price
- **TDM interconnect** for trunking and analog connections, including fax machines, lobby phones, etc.
- **Get started quickly**–both the SBC 2000 and SBC 1000 can be provisioned and operational in as little as one hour
- **Microsoft Lync quality of experience (QoE) monitoring**–the SBC 1000 and SBC 2000 are the only SBCs on the market that monitor the entire call flow

(Source :

http://www.exertisgoconnect.nl/products/images/files/brochure_Sonus_SBC_Portfolio.pdf)

Sonus SBC 2000™ Session Border Controller

The Sonus SBC 2000 Session Border Controller is an advanced SBC designed to help medium-sized enterprise networks safely and cost-effectively embrace the new multi-vendor world of SIP-based communications, such as Voice over IP (VoIP) and Unified Communications. The SBC 2000 delivers all of the features you would expect in an enterprise-class SBC, including security, built-in media transcoding, SIP interworking and intelligent call routing. The Sonus SBC 2000 also has a unique feature not found in other solutions: survivable branch appliance (SBA) functionality that enables the SBC to complete voice calls over the PSTN should the enterprise WAN go down.

Optimized for Lync

Features/Benefits

- **Low cost of entry**–session-based licensing allows enterprises to start small and scale as they grow (up to 600 concurrent sessions) without high upfront costs
- **Performance you can trust**–combines SBA features in an SBC device for 24/7/365 communications even during IP network outages
- **TDM interconnect** for trunking and analog connections, including fax machines, lobby phones, etc.
- **Microsoft Lync quality of experience (QoE) monitoring**–the SBC 1000 and SBC 2000 are the only SBCs on the market that monitor the entire call flow

(Source :

http://www.exertisgoconnect.nl/products/images/files/brochure_Sonus_SBC_Portfolio.pdf)

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SBC 1000/2000 5.0.x

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Configuring Quality of Experience (QoE) Settings

Note
QoE Reporting must be enabled in the SIP Channels and Routing Panel of the relevant SIP Signaling group before statistics can be collected.

Related Documents
- Configuring the QoE Feature

Overview

The SBC Quality of Experience (QoE) feature collects data about call RTP streams, generates MOS information, and sends them to a MS Lync QoE server.

For more information about RTCP and MOS see the following:

- RFC 3611
- Mean Opinion Score and Metrics

(Source :

https://support.sonus.net/display/UXDOC50/Configuring+Quality+of+Experience+%28QoE%29+Settings)

The Ribbon Communications SBC 1000™ Gateway

Protocol Support

- SNMPv2c, SNMPv3
- HTTPS
- SIP (RFC 3261) over UDP, TCP, TLS
- RTP/RTCP (RFC 3550, 3551)
- RTP/RTCP multiplexing over single UDP port (RFC 5761)
- DNS
- IPv4, IPv6, and IPv4/IPv6 interworking
- RIPv2, OSPF as dynamic IP routing protocols
- DHCP server
- DHCP client
- Asynchronous DNS for SIP
- NAT
- Support for Reason Header

(Source :

https://www.voipsupply.com/downloads/dl/file/id/37491/sbc_1000_gateway_datasheet.pdf)

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Protocol Support

- SNMPv2c, SNMPv3
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- DHCP server
- DHCP client
- Asynchronous DNS for SIP
- NAT
- Support for Reason Header

(Source : https://www.voipsupply.com/downloads/dl/file/id/37471/sbc_2000_datasheet.pdf)

```
RFC 3611                        RTCP XR                    November 2003

References

Normative References

   [1]  Bradner, S., "Key words for use in RFCs to Indicate Requirement
        Levels", BCP 14, RFC 2119, March 1997.

   [2]  Crocker, D., Ed. and P. Overell, "Augmented BNF for Syntax
        Specifications: ABNF", RFC 2234, November 1997.

   [3]  ETSI, "Quality of Service (QoS) measurement methodologies", ETSI
        TS 101 329-5 V1.1.1 (2000-11), November 2000.

   [4]  Handley, M. and V. Jacobson, "SDP: Session Description
        Protocol", RFC 2327, April 1998.

   [5]  Hovey, R. and S. Bradner, "The Organizations Involved in the
        IETF Standards Process", BCP 11, RFC 2028, October 1996.

   [6]  ITU-T, "The E-Model, a computational model for use in
        transmission planning", Recommendation G.107, January 2003.

   [7]  Narten, T. and H. Alvestrand, "Guidelines for Writing an IANA
        Considerations Section in RFCs", BCP 26, RFC 2434, October 1998.
```

(Source : https://tools.ietf.org/html/rfc3611)

```
   MOS-CQ: 8 bits
         The estimated mean opinion score for conversational quality
         (MOS-CQ) is defined as including the effects of delay and other
         effects that would affect conversational quality.  The metric
         may be calculated by converting an R factor determined
         according to ITU-T G.107 [6] or ETSI TS 101 329-5 [3] into an
         estimated MOS using the equation specified in G.107.  It is
         expressed as an integer in the range 10 to 50, corresponding to
         MOS x 10, as for MOS-LQ.

         A value of 127 indicates that this parameter is unavailable.
         Values other than 127 and the valid range defined above MUST
         not be sent and MUST be ignored by the receiving system.
```

(Source : https://tools.ietf.org/html/rfc3611)

49. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 1 of the '230 Patent. Ribbon's infringement in this regard is ongoing.

50. Ribbon has infringed the '230 Patent by using the accused products and thereby practicing a method for determining acceptability of quality of a second communications service, in comparison to a first communications service which is deemed to exhibit acceptable quality.

For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The ITU-T G.107 Recommendation includes an E-model for calculating voice quality as perceived by a typical telephone user. The E-model outputs a transmission rating factor i.e., R, which can be transformed into Mean Opinion Score—i.e., MOS value—that represents the voice quality. This E-model is applied to a real-time voice call ("second communication service") for measuring its voice quality. The E-model is based on modeling the results from multiple subjective tests performed on a wide range of transmission parameters. It also includes a reference table with different R value and MOS value thresholds, and corresponding perceived voice quality. The MOS values (quality index) in the reference table are obtained using an aggregate of data from multiple test calls using a first communication service. A MOS value of 4.34 and above, or an R value of 90 and above is considered to be very satisfied ("acceptable quality").

The E-model: a computational model for use in transmission planning

1 Scope

This Recommendation describes a computational model, known as the E-model, that has proven useful as a transmission planning tool for assessing the combined effects of variations in several transmission parameters that affect the conversational[1] quality of 3.1 kHz handset telephony. This computational model can be used, for example, by transmission planners to help ensure that users will be satisfied with end-to-end transmission performance whilst avoiding over-engineering of networks. It must be emphasized that the primary output from the model is the "rating factor" *R* but this can be transformed to give estimates of customer opinion. Such estimates are only made for transmission planning purposes and not for actual customer opinion prediction (for which there is no agreed-upon model recommended by the ITU-T).

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the R-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

The E-model (**ITU-T Rec. G.107** [1]) is a transmission planning tool that provides a prediction of the expected voice quality, as perceived by a typical telephone user, for a complete end-to-end (i.e. mouth-to-ear) telephone connection under conversational conditions. The E-model takes into account a wide range of telephony-band impairments, in particular the impairment due to low bit-rate coding devices and one-way delay, as well as the "classical" telephony impairments of loss, noise and echo. It can be applied to assess the voice quality of wireline and wireless scenarios, based on circuit-switched and packet-switched technology.

The E-model is based on modeling the results from a large number of subjective tests done in the past on a wide range of transmission parameters. The primary output of the E-model calculations is a scalar quality rating value known as the "Transmission Rating Factor, R". R can be transformed into other quality measures such as Mean Opinion Score (MOS-CQE [2]), Percentage Good or Better ((GoB) or Percentage Poor or Worse ((PoW). However, caution should be exercised when comparing these transformed measures with values of MOS, %GoB or %PoW from other sources, which may not have been obtained under comparable conditions.

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

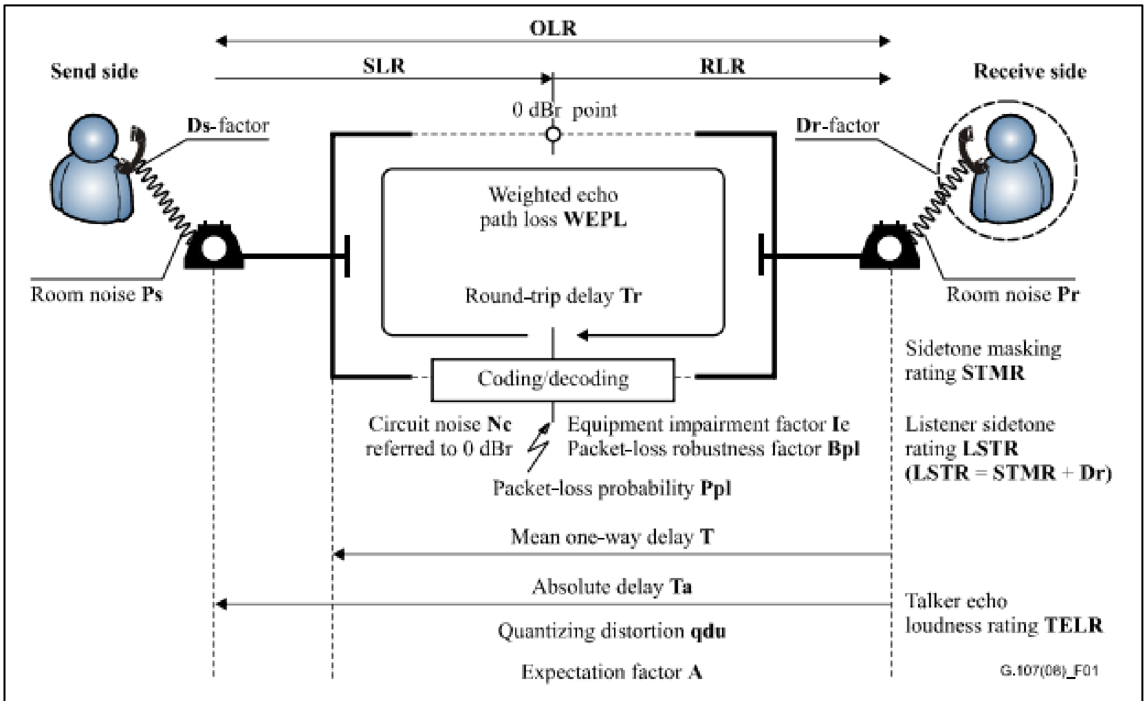

Figure 1 – Reference connection of the E-model

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

***R*-value (lower limit)**	**MOS_{CQE} (lower limit)**	**GoB (%) (lower limit)**	**PoW (%) (upper limit)**	**User satisfaction**
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

51. The methods practiced by Ribbon's use of the accused products include obtaining a first quality index pertaining to the first communications service. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The E-model includes a reference table with different R value and MOS value thresholds, and corresponding

perceived voice quality. Further, the E-model is modelled using large number of subjective tests. Thus, the thresholds of MOS value are based on an aggregate of multiple subjective calls' data. For instance, a MOS value ("first quality index") threshold of 4.34 for best quality is based on an aggregate of data for multiple high quality voice calls using a first communication service.

The E-model (**ITU-T Rec. G.107** [1]) is a transmission planning tool that provides a prediction of the expected voice quality, as perceived by a typical telephone user, for a complete end-to-end (i.e. mouth-to-ear) telephone connection under conversational conditions. The E-model takes into account a wide range of telephony-band impairments, in particular the impairment due to low bit-rate coding devices and one-way delay, as well as the "classical" telephony impairments of loss, noise and echo. It can be applied to assess the voice quality of wireline and wireless scenarios, based on circuit-switched and packet-switched technology.

The E-model is based on modeling the results from a large number of subjective tests done in the past on a wide range of transmission parameters. The primary output of the E-model calculations is a scalar quality rating value known as the "Transmission Rating Factor, R". R can be transformed into other quality measures such as Mean Opinion Score (MOS-CQE [2]), Percentage Good or Better ((GoB) or Percentage Poor or Worse ((PoW). However, caution should be exercised when comparing these transformed measures with values of MOS, %GoB or %PoW from other sources, which may not have been obtained under comparable conditions.

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the *R*-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6}$ (B-4)

For $R > 100$: $MOS_{CQE} = 4.5$

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60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

Table 1 - Definition of categories of speech transmission quality

Range of E-model Rating R	Speech transmission quality category	User satisfaction
$90 \le R < 100$	Best	Very satisfied
$80 \le R < 90$	High	Satisfied
$70 \le R < 80$	Medium	Some users dissatisfied
$60 \le R < 70$	Low	Many users dissatisfied
$50 \le R < 60$	Poor	Nearly all users dissatisfied

NOTE 1 – Connections with E-model Ratings R below 50 are not recommended.
NOTE 2 – Although the trend in transmission planning is to use E-model Ratings R, equations to convert E-model Ratings R into other metrics, e.g. %MOS, %GoB, PoW can be found in **ITU-T Rec. G.107 Annex B** [1].

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

52. The methods practiced by Ribbon's use of the accused products include obtaining a second quality index pertaining to the second communications service. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The E-model is applied to a real-time voice call made using a system including an accused product ("second communication service") for measuring its voice quality by calculating the R value and its corresponding MOS value ("second quality index").

7.1 Calculation of the transmission rating factor, R

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model, see [b-ITU-T P-Sup.3].

Psychological factors on the psychological scale are additive.

The result of any calculation with the E-model in a first step is a transmission rating factor *R*, which combines all transmission parameters relevant for the considered connection. This rating factor *R* is composed of:

$$R = Ro - Is - Id - Ie\text{-}eff + A \qquad (7\text{-}1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. Factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie-eff* represents impairments caused by low bit-rate codecs. It also includes impairment due to randomly distributed packet losses. The advantage factor *A* allows for compensation of impairment factors when the user benefits from other types of access to the user. The term *Ro* and the *Is* and *Id* values are subdivided into further specific impairment values. The following clauses give the equations used in the E-model.

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

An estimated mean opinion score (MOS_{CQE}) for the conversational situation in the scale 1-5 can be obtained from the R-factor using the equations:

For $R < 0$: $MOS_{CQE} = 1$

For $0 < R < 100$: $$MOS_{CQE} = 1 + 0.035R + R(R-60)(100-R)7 \cdot 10^{-6} \quad \text{(B-4)}$$

For $R > 100$: $MOS_{CQE} = 4.5$

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

53. The methods practiced by Ribbon's use of the accused products include determining that the second communication service is of unacceptable quality if the second quality index differs from the first quality index service by more than a selected amount. For example, the accused products are used by Ribbon to implement the ITU-T G.107 Recommendation. The calculated MOS value is then compared with the reference table to determine the perceived voice quality. If the R value differs from a R value of 90 by more than 40, then the call is considered to be of unacceptable quality. Similarly, if the calculated MOS value ("second quality index") differs from a MOS value of 4.34 ("first quality index") by more than 1.76, then the call is considered to be of unacceptable quality.

Table B.1 – Provisional guide for the relation between *R*-value and user satisfaction

***R*-value (lower limit)**	**MOS_{CQE} (lower limit)**	**GoB (%) (lower limit)**	**PoW (%) (upper limit)**	**User satisfaction**
90	4.34	97	~0	Very satisfied
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50	2.58	27	38	Nearly all users dissatisfied

(Source: https://www.itu.int/rec/dologin_pub.asp?lang=s&id=T-REC-G.107-201402-S!!PDF-E&type=items)

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80 ≤ R < 90	High	Satisfied
70 ≤ R < 80	Medium	Some users dissatisfied
60 ≤ R < 70	Low	Many users dissatisfied
50 ≤ R < 60	Poor	Nearly all users dissatisfied

NOTE 1 – Connections with E-model Ratings R below 50 are not recommended.
NOTE 2 – Although the trend in transmission planning is to use E-model Ratings R, equations to convert E-model Ratings R into other metrics, e.g. %MOS, %GoB, PoW can be found in **ITU-T Rec. G.107 Annex B** [1].

(Source: https://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm)

54. Far North Patents only asserts method claims from the ‘230 Patent.

55. Ribbon has had knowledge of the ‘230 Patent at least as of the date when it was notified of the filing of this action.

56. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

57. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the ‘230 Patent.

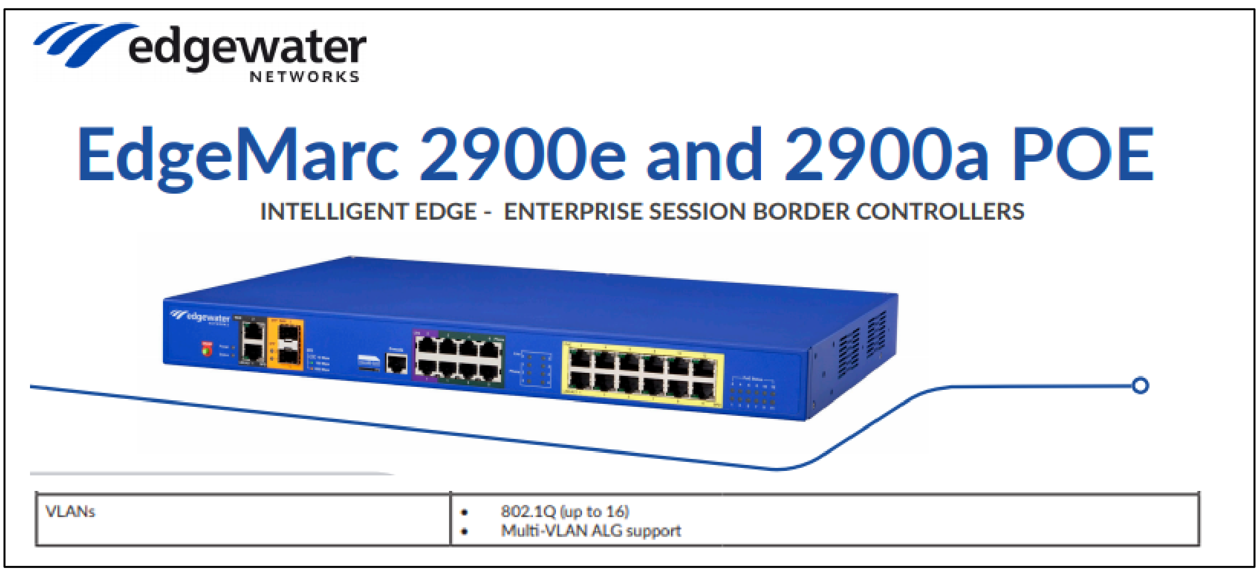
COUNT IV

DIRECT INFRINGEMENT OF U.S. PATENT NO. 8,306,053

58. On November 6, 2012, United States Patent No. 8,306,053 ("the '053 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Methods and Apparatus for Providing Quality-of-Service Guarantees in Computer Networks."

59. Far North Patents is the owner of the '053 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '053 Patent against infringers, and to collect damages for all relevant times.

60. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Ribbon/Edgewater EdgeMarc 2900 enterprise session border controller family of products that include advanced quality of service capabilities (collectively, "accused products").



(Source : https://www.teledynamics.com/tdresources/d82d2f8c-7cbd-44dc-aa37-68d5b81a3ef8.pdf)

Edgewater Networks is Now Ribbon

The full suite of the Edgewater solutions and products are available from Ribbon.

(Source : https://info.rbbn.com/edgewater/)

Edgewater Networks Announces Launch of the EdgeMarc 2900 Series

June 06, 2017 12:00 PM Eastern Daylight Time

SAN JOSE, Calif.--(BUSINESS WIRE)--Edgewater Networks, the market leader in Network Edge Orchestration, announced the availability of the EdgeMarc 2900 Intelligent Edge devices. The EdgeMarc 2900 series enables enterprises and service providers to future-proof their SIP trunking and Unified Communications deployments, providing a highly flexible, scalable, and secure platform for service delivery and ongoing service quality management.

EdgeMarc 2900 Intelligent Edge breaks new ground in #NetworkEdgeOrchestration to future-proof SIP trunking and UC.

Tweet this

The EdgeMarc 2900e offers dual Ethernet and Optical WAN connections to extend the capabilities of Edgewater Networks' Network Edge Orchestration platform and align with Edgewater Networks' SD-WAN solution. The EdgeMarc 2900e supports WAN connections up to 1 Gbps and can be used for Small and Medium Business applications or Mid-Market deployments up to 300 concurrent calls. "We're excited to announce the availability of the first Intelligent Edge in our EdgeMarc 2900 series that enables service providers to deliver optimized VoIP solutions to nearly every use case in the market," said Chris Kolstad, VP of Product Management at Edgewater Networks. "The inherent flexibility and power of the EdgeMarc 2900 enables deployments in SMB environments with as few as five seats to enterprise environments with thousands of seats."

Future Intelligent Edge products will include the EdgeMarc 2900a and the EdgeMarc 2900POE. The EdgeMarc 2900a will add both outbound and inbound analog line (FXO/FXS) support for applications such as PSTN fail-over, fax, and overhead paging. The EdgeMarc 2900POE will integrate managed POE ports to provide an all-in-one solution for small office implementations.

(Source : https://www.businesswire.com/news/home/20170606005340/en/Edgewater-Networks-Announces-Launch-EdgeMarc-2900-Series)

61. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claims 1 and 14 of the '053 Patent. Ribbon's infringement in this regard is ongoing.

62. Ribbon has infringed the '053 Patent by making, having made, using, importing, providing, supplying, distributing, selling or offering for sale products including a device adapter comprising a transmission unit configured to transmit data from a real time device via a network according to a time frame, wherein the time frame is substantially synchronized in the device adapter and at least one other device adapter, the time frame repeating periodically and including a plurality of assigned time phases and a free access phase. For example, the accused products are configured to be used to implement the IEEE 802.1Q standard. IEEE standard 802.1Q implements a method in which a time aware bridged LAN transmits data from one end point i.e, Precision Time Protocol ("PTP") instance to another. The endpoints transmit data that is a mix of time-critical traffic and other traffic, (i.e, real time data and non-real time data) via PTP instances such as bridges (device adapters). The end point can be either a real time device or a non-real time device. The bridged network uses 802.1AS base time to synchronize all the clocks of ports associated with bridges (device adapters). Using the Best Master Clock Algorithm, the synchronization time signal is transmitted from a grandmaster to other ports. IEEE Std. 802.1AS™-2011 is normative and essential to implement an IEEE Std. 802.1Q Compliant System. IEEE Std. 802.1Q-2018 defines parameters, such as AdminBaseTime and OpenBaseTime, which are used to synchronize the clocks across the network. The bridges containing ports schedule the transmission of traffic based on the synchronized time. Each port associated with a specific set of transmission queues includes a plurality of transmission gates. A transmission gate can be in a closed state or open state. The functionality of assigned time phases

is achieved using open gates transmitting data packets during transmission time. The functionality of a free access phase is achieved when the gates are opened for transmission during any time. IEEE Std. 802.1Q-2018 supports cyclic queuing and forwarding structures to create synchronized frames and gates which repeat periodically (Annex T).

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

5.2.b. Scope of the project: This amendment specifies time-aware queue-draining procedures, managed objects and extensions to existing protocols that enable bridges and end stations to schedule the transmission of frames based on a synchronized time. Virtual Local Area Network (VLAN) tag encoded priority values are allocated allowing simultaneous support of scheduled traffic, credit-based shaper traffic and other bridged traffic over Local Area Networks (LANs).

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

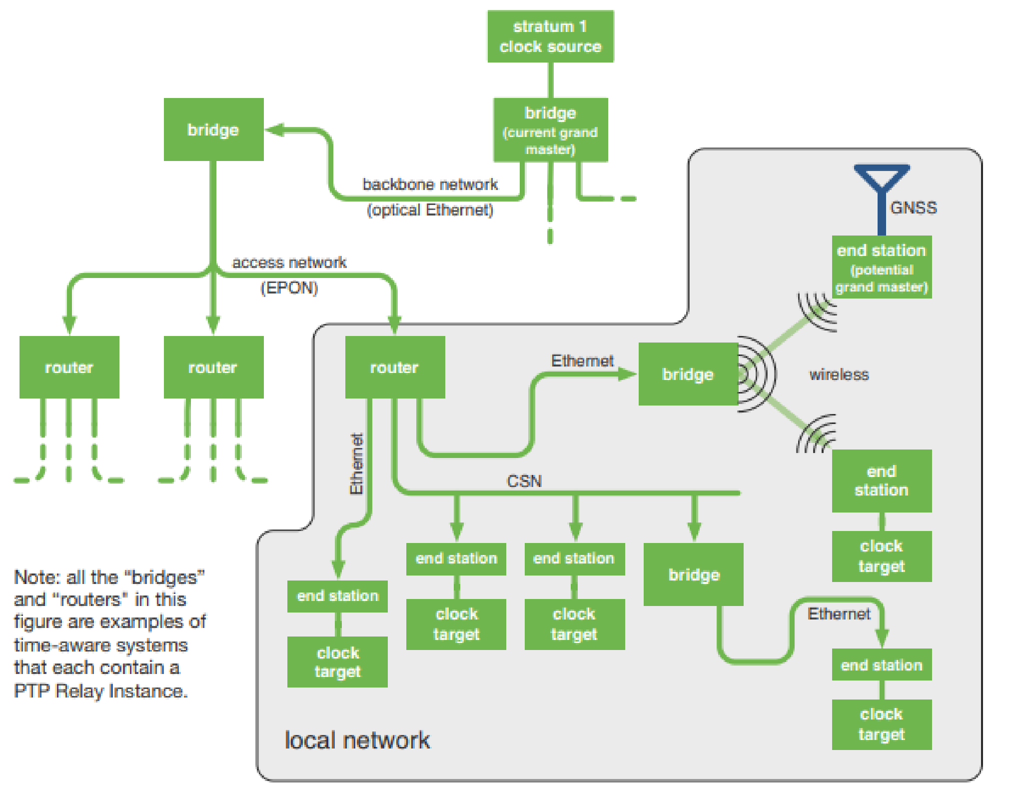


Figure 7-1—Time-aware network example

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

7.2 Architecture of a time-aware network

7.2.1 General

A time-aware network consists of a number of interconnected time-aware systems that support the gPTP defined within this standard. These time-aware systems can be any networking device, including, for example, bridges, routers, and end stations. A set of time-aware systems that are interconnected by gPTP-capable network elements is called a *gPTP network*. Each instance of gPTP that the time-aware systems support is in one *gPTP domain*, and the instances of gPTP are said to be part of that gPTP domain. A time-aware system can support, and therefore be part of, more than one gPTP domain. The entity of a single time-aware system that executes gPTP in one gPTP domain is called a PTP Instance. A time-aware system can contain multiple PTP Instances, which are each associated with a different gPTP domain. There are two

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

3.16 PTP End Instance: A PTP Instance that is capable of acting as the source of synchronized time on the network, or destination of synchronized time using the IEEE 802.1AS protocol, or both.

3.17 PTP Instance: A PTP Relay Instance or a PTP End Instance. A PTP Instance implements those portions of this standard indicated as applicable to a PTP Relay Instance or a PTP End Instance. Each PTP Instance operates in exactly one domain.

3.18 PTP Link: Within a domain, a network segment between two PTP Ports using the peer-to-peer delay mechanism of this standard. The peer-to-peer delay mechanism is designed to measure the propagation time over such a link.

NOTE—A PTP Link between PTP Ports of PTP Instances is also a gPTP Communication Path (see 3.9).

3.19 PTP Relay Instance: A PTP Instance that is capable of communicating synchronized time received on one port to other ports, using the IEEE 802.1AS protocol.

NOTE—A PTP Relay Instance could, for example, be contained in a bridge, a router, or a multi-port end station.

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

Traffic scheduling

This annex provides some background to the mechanisms provided in this standard for traffic scheduling, with the intent of providing some insight into the motivation for the provision of mechanisms for traffic scheduling and some of the ways that the mechanisms might be used.

Q.1 Motivation

Some applications have a need for frame delivery that is highly predictable in terms of the time at which frame transmission will occur, and the overall latency and jitter that will be experienced as the frame is propagated to its destination. Examples include industrial and automotive control applications, where data transmitted over the network is used to feed the parameters of control loops that are critical to the operation of the plant or machinery involved, and where frames carrying control data are transmitted on a repeating time schedule; late delivery of such frames can result in instability, inaccuracy, or failure of the operation of the control loops concerned. In some implementations, this need has been met by the provision of dedicated, highly engineered networks that are used solely for the transmission of time-critical control traffic; however, as the bandwidth occupied by such traffic is often low, and the cost of providing a dedicated control network can be high, it can be desirable to mix time-critical traffic with other classes of traffic in the same network, as long as such mixing can be achieved while still meeting the timing requirements of the time-critical traffic.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

b) gPTP specifies a media-independent sublayer that simplifies the integration within a single timing domain of multiple different networking technologies with radically different media access protocols. gPTP specifies a media-dependent sublayer for each medium. The information exchanged between PTP Instances has been generalized to support different packet formats and management schemes appropriate to the particular networking technology. IEEE Std 1588-2019, on the other

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

7.2.2 Time-aware network consisting of a single gPTP domain

Figure 7-1 illustrates an example time-aware network consisting of a single gPTP domain, using all theabove network technologies (i.e., (c) - (f) of 7.2.1), where end stations on several local networks are connected to a grandmaster on a backbone network via an EPON access network.

Any PTP Instance with clock sourcing capabilities can be a potential grandmaster, so there is a selection method (the *best master clock algorithm*, or BMCA) that ensures that all of the PTP Instances in a gPTP domain use the same grandmaster.[12] The BMCA is largely identical to that used in IEEE Std 1588-2019,

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

10.3.1.1 Best master clock algorithm overview

In the BMCA (i.e., method (a) of 10.3.1), best master selection information is exchanged between PTP Instances of time-aware systems via Announce messages (see 10.5 and 10.6). Each Announce message contains time-synchronization spanning tree vector information that identifies one PTP Instance as the root of the time-synchronization spanning tree and, if the PTP Instance is grandmaster-capable, the grandmaster. Each PTP Instance in turn uses the information contained in the Announce messages it receives, along with its knowledge of itself, to compute which of the PTP Instances that it has knowledge of ought to be the root of the spanning tree and, if grandmaster-capable, the grandmaster. As part of constructing the time-synchronization spanning tree, each port of each PTP Instance is assigned a port state from Table 10-2 by state machines associated with the ports and with the PTP Instance as a whole.

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

c) In gPTP there are only two types of PTP Instances: PTP End Instances and PTP Relay Instances, while IEEE 1588 has ordinary clocks, boundary clocks, end-to-end transparent clocks, and P2P transparent clocks. A PTP End Instance corresponds to an IEEE 1588 ordinary clock, and a PTP Relay Instance is a type of IEEE 1588 boundary clock where its operation is very tightly defined, so much so that a PTP Relay Instance with Ethernet ports can be shown to be mathematically equivalent to a P2P transparent clock in terms of how synchronization is performed, as shown in

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

- **Grand Master selection**
  - GM-capable stations advertise themselves via ANNOUNCE messages
  - If a station hears from station with "better" clock, it does not send ANNOUNCE
    - Configurable "Priority" field can override clock quality
    - MAC address is tie breaker
  - Time relays drop all inferior ANNOUNCE messages
    - Forward only the best
  - Last one standing is Grand Master for the domain
    - GM is the root of the 802.1AS timing tree
    - GM periodically sends the current time

(Source: https://avnu.org/wp-content/uploads/2014/05/as-kbstanton-8021AS-tutorial-0714-v01.pdf)

IEEE Std 802.1Q-2018
IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks

2. Normative references

The following referenced documents are indispensable for the application of this document (i.e., they must be understood and used, so each referenced document is cited in the text and its relationship to this document is explained). For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies.

ANSI X3.159, American National Standards for Information Systems—Programming Language—C.[5]

IEEE Std 802®, IEEE Standard for Local and metropolitan area networks: Overview and Architecture.[6,7]

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IEEE Std 802.1AC™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Service Definition.

IEEE Std 802.1AE™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Security.

IEEE Std 802.1AS™, IEEE Standard for Local and metropolitan area networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

8.6.9.4.1 AdminBaseTime

The administrative value of base time, expressed as an IEEE 1588 precision time protocol (PTP) [B24] timescale (see 8.2 of IEEE Std 802.1AS™-2011 [B11]). This value can be changed by management, and is used by the List Config state machine (8.6.9.3) to set the value of OperBaseTime (8.6.9.4.18).

NOTE—Time is expressed in the PTP timescale as the number of seconds, nanoseconds, and fractional nanoseconds that have elapsed since 1 January 1970 00:00:00 TAI.

NOTE 2—If AdminBaseTime is set to the same time in the past in all bridges and end stations, OperBaseTime is always in the past, and all cycles start synchronized. Using AdminBaseTime in the past is appropriate when you can start schedules prior to starting the application that uses the schedules. Use of AdminBaseTime in the future is intended to change a currently running schedule in all bridges and end stations to a new schedule at a future time. Using AdminBaseTime in the future is appropriate when schedules must be changed without stopping the application.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

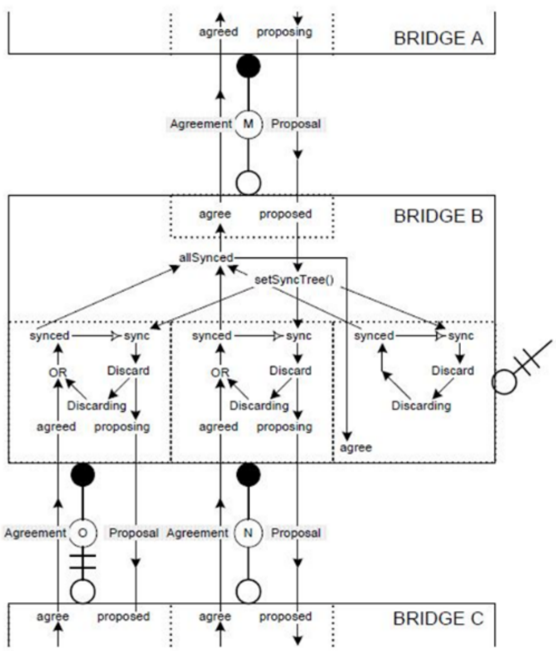

Figure 13-10—Agreements and Proposals

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

> NOTE 2—Agreements can be generated without prior receipt of a Proposal as soon as the necessary conditions are met. Subsequent receipt of a Proposal serves to elicit a further Agreement. If all other ports have already been synchronized (allSynced in Figure 13-10) and the Proposal's priority vector does not convey worse information, synchronization is maintained and there is no need to transition Designated Ports to Discarding once more, or to transmit further Proposals.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

> **8.6.8.4 Enhancements for scheduled traffic**
>
> A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:
>
> a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
> b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

> **8.6.8 Transmission selection**
>
> For each Port, frames are selected for transmission on the basis of the traffic classes that the Port supports and the operation of the transmission selection algorithms supported by the corresponding queues on that Port. For a given Port and traffic class, frames are selected from the corresponding queue for transmission if and only if
>
> a) The operation of the transmission selection algorithm supported by that queue determines that there is a frame available for transmission; and
>
> b) For each queue corresponding to a numerically higher value of traffic class supported by the Port, the operation of the transmission selection algorithm supported by that queue determines that there is no frame available for transmission.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

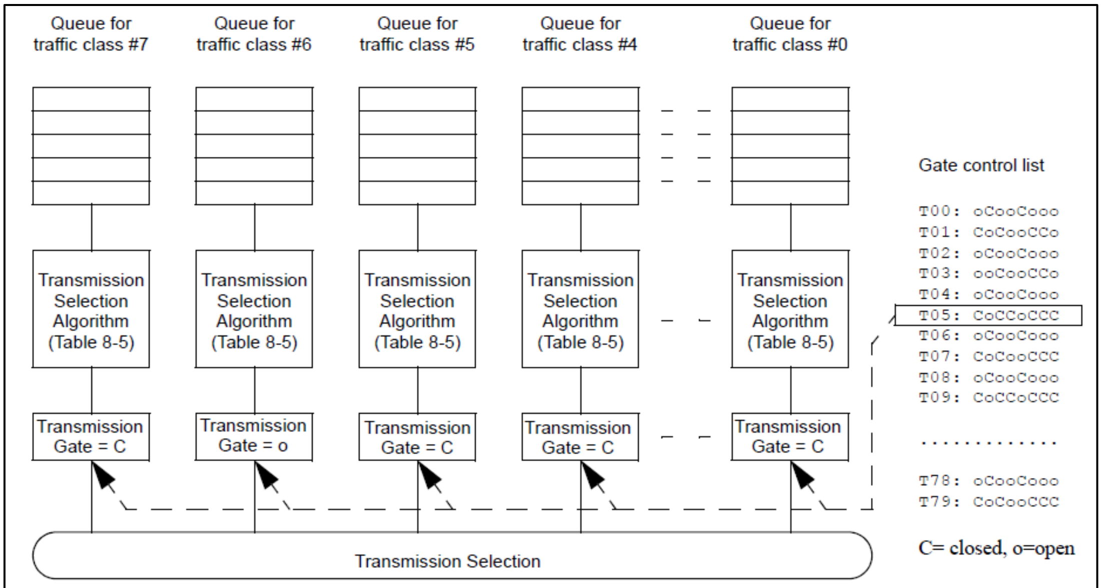

Figure 8-12—Transmission selection with gates

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

> A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015)).

> **3.3 gating cycle:** The period of time over which the sequence of operations in a gate control list repeats.
>
> **3.4 transmission gate:** A gate that connects or disconnects the transmission selection function of the forwarding process from the queue, allowing or preventing it from selecting frames from that queue. The gate has two states, Open and Closed.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

Cyclic queuing and forwarding[53]

T.1 Overview of CQF

Cyclic queuing and forwarding (CQF) is a method of traffic shaping that can deliver deterministic, and easily calculated, latency for time-sensitive traffic streams. As the name implies, the principle underlying CQF is that stream traffic is transmitted and queued for transmission along a network path in a cyclic manner. Time is divided into numbered time intervals i, $i+1$, $i+2$, ... $i+N$, each of duration d. Frames transmitted by a Bridge, *Alice*, during time interval i are received by a downstream Bridge, *Bob*, during time interval i and are transmitted onwards by *Bob* towards Bridge *Charlie* during time interval $i+1$, and so on. A starting assumption is that, for a given traffic class, all Bridges and all end stations connected to a given bridge have a common understanding (to a known accuracy) of the start time of cycle i, and the cycle duration, d.

Frames transmitted by *Alice* during interval i are transmitted by *Bob* in interval $i+1$; the maximum possible delay experienced by a given frame is from the beginning of i to the end of $i+1$, or twice d. Similarly, the minimum possible delay experienced is from the end of i to the beginning of $i+1$, which is zero. More generally, the maximum delay experienced by a given frame is

$$(h+1) \times d$$

and the minimum delay experienced by a given frame is

$$(h-1) \times d$$

where h is the number of hops.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

63. The accused products include a device adapter that is configured to transmit data during at least one of an assigned time phase associated with the device adapter prior to transmission of data from the real-time device by the device adapter, and included in the plurality of assigned time phases, or the free access phase, to refrain from transmitting data during time phases of the plurality of assigned time phases that are not associated with the device adapter, and to be able to determine whether to defer transmission of data during the assigned time phase associated with the device adapter and the free access phase to allow a non-real time device to transmit data. For example, the accused products are configured to be used to implement the IEEE 802.1Q standard. IEEE standard 802.1Q shows that scheduling of ports' transmission gates (configured) for transmission of data starts prior to the transmission of real-time data. The

functionality of assigned time phases is achieved using open gates transmitting data packets during scheduled transmission time. Per Clause 8.6.8 of the IEEE Std. 802.1Q, each time phase is assigned to a specific device adapter prior to transmission of real-time data by the specific device adapter. Furthermore, IEEE 802.1Q performs traffic shaping through Per-Stream Filtering and Policing (PSTP). IEEE standard 802.1Q implements a method in which open gates transmit data packets during transmission time and closed gates refrain data packets from transmission. The functionality of assigned time phases is achieved using open gates transmitting data packets during transmission time. The functionality of free access phase is achieved by the gates that are opened for transmission during any time. IEEE Std. 802.1Q supports Forwarding and Queuing Enhancements for Time Sensitive Streams. Thus, one of the plurality of device adaptors is configured to transmit data during at least one of a respective assigned time phase or free access phase, to refrain from transmitting data during time phases not assigned to the respective one of the plurality of device adapters. IEEE Std. 802.1Q provides traffic shaping for various classes of data transmission and determining whether to defer transmission of data during at least one of the assigned time phase or the free access phase to allow a non-real time device to transmit data.

NOTE 2—If AdminBaseTime is set to the same time in the past in all bridges and end stations, OperBaseTime is always in the past, and all cycles start synchronized. Using AdminBaseTime in the past is appropriate when you can start schedules prior to starting the application that uses the schedules. Use of AdminBaseTime in the future is intended to

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

5.5 Need for the Project: The credit-based shaper works well in arbitrary networks (i.e., non-engineered). Networks employing scheduled transmissions are able to control real-time processes. This amendment enables those two kinds of networks to be consolidated into a single network, with a significant cost reduction to the user.

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

8.6.8 Transmission selection

For each Port, frames are selected for transmission on the basis of the traffic classes that the Port supports and the operation of the transmission selection algorithms supported by the corresponding queues on that Port. For a given Port and traffic class, frames are selected from the corresponding queue for transmission if and only if

a) The operation of the transmission selection algorithm supported by that queue determines that there is a frame available for transmission; and

b) For each queue corresponding to a numerically higher value of traffic class supported by the Port, the operation of the transmission selection algorithm supported by that queue determines that there is no frame available for transmission.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

5.4.1.9 Cyclic queuing and forwarding (CQF) requirements

A VLAN Bridge component implementation that conforms to the provisions of this standard for CQF (see Annex T) shall

a) Support the enhancements for scheduled traffic as specified in 8.6.8.4.

b) Support the state machines for scheduled traffic as specified in 8.6.9.
c) Support the state machines for stream gate control as specified in 8.6.10.
d) Support the management entities for scheduled traffic as specified in 12.29.
e) Support the requirements for per-stream filtering and policing (PSFP) as stated in 5.4.1.8.
f) Support the management entities for PSFP as specified in 12.31.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

5.4.1.5 Forwarding and Queuing Enhancements for time-sensitive streams (FQTSS)—requirements

A VLAN Bridge component implementation that conforms to the provisions of this standard for FQTSS shall

a) Support a minimum of two traffic classes on all Ports, of which
   1) A minimum of one traffic class supports the strict priority algorithm for transmission selection (8.6.8.1), and
   2) One traffic class is a stream reservation (SR) class.
b) Support the operation of the credit-based shaper algorithm (8.6.8.2) on all Ports as the transmission selection algorithm used for the SR class.
c) Support SRP domain boundary port priority regeneration override as defined in 6.9.4, and the default priority regeneration override value defined in Table 6-5, for SR class "B."
d) Support the tables and procedures for mapping priorities to traffic classes as defined in 34.5.

SR: Stream Reservation, SRP: Stream Reservation Protocol

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

8.6.8.4 Enhancements for scheduled traffic

A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:

a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

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d) Support the tables and procedures for mapping priorities to traffic classes as defined in 34.5.

SR: Stream Reservation, SRP: Stream Reservation Protocol

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

5.4.1.6 ETS Bridge requirements

A device supporting ETS shall

a) Support at least 3 traffic classes (37.3).

NOTE—A minimum of 3 traffic classes allows a minimum configuration such that one traffic class contains priorities with PFC enabled, one traffic class contains priorities with PFC disabled, and one traffic class using strict priority.

b) Support bandwidth configuration with a granularity of 1% or finer (37.3).

c) Support bandwidth allocation with a precision of 10% (37.3).

d) Support a transmission selection policy such that if one of the traffic classes does not consume its allocated bandwidth, then any unused bandwidth is available to other traffic classes (37.3).

e) Support DCBX (Clause 38).

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

64. Ribbon has infringed the '053 Patent by using the accused products and thereby practicing a method that includes transmitting a synchronization signal at regular intervals to synchronize local clocks of each of a plurality of device adapters. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. IEEE standard 802.1Q implements a method in which a time aware bridged LAN use 802.1AS base time to synchronize

all the clocks of ports associated with bridges (device adapters) by periodically (regular intervals) sending the Announce messages. Using Best Master Clock Algorithm, the synchronization time signal is transmitted from a grandmaster to other ports via periodic Announce messages. IEEE Std. 802.1AS™-2011 is normative and essential to implement an IEEE Std. 802.1Q Compliant System. IEEE Std. 802.1Q-2018 defines parameters, such as AdminBaseTime and OpenBaseTime, which are used to synchronize the clocks across the network.

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

5.2.b. Scope of the project: This amendment specifies time-aware queue-draining procedures, managed objects and extensions to existing protocols that enable bridges and end stations to schedule the transmission of frames based on a synchronized time. Virtual Local Area Network (VLAN) tag encoded priority values are allocated allowing simultaneous support of scheduled traffic, credit-based shaper traffic and other bridged traffic over Local Area Networks (LANs).

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

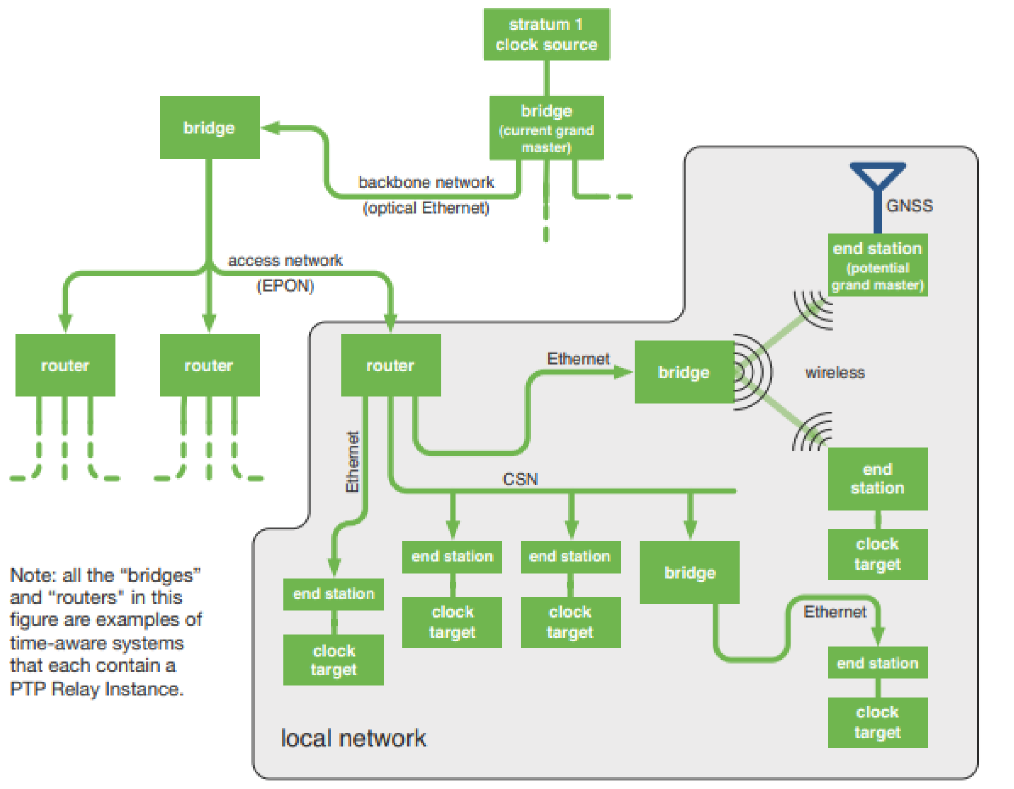

Figure 7-1—Time-aware network example

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

7.2 Architecture of a time-aware network

7.2.1 General

A time-aware network consists of a number of interconnected time-aware systems that support the gPTP defined within this standard. These time-aware systems can be any networking device, including, for example, bridges, routers, and end stations. A set of time-aware systems that are interconnected by gPTP-capable network elements is called a *gPTP network*. Each instance of gPTP that the time-aware systems support is in one *gPTP domain*, and the instances of gPTP are said to be part of that gPTP domain. A time-aware system can support, and therefore be part of, more than one gPTP domain. The entity of a single time-aware system that executes gPTP in one gPTP domain is called a PTP Instance. A time-aware system can contain multiple PTP Instances, which are each associated with a different gPTP domain. There are two

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

3.16 PTP End Instance: A PTP Instance that is capable of acting as the source of synchronized time on the network, or destination of synchronized time using the IEEE 802.1AS protocol, or both.

3.17 PTP Instance: A PTP Relay Instance or a PTP End Instance. A PTP Instance implements those portions of this standard indicated as applicable to a PTP Relay Instance or a PTP End Instance. Each PTP Instance operates in exactly one domain.

3.18 PTP Link: Within a domain, a network segment between two PTP Ports using the peer-to-peer delay mechanism of this standard. The peer-to-peer delay mechanism is designed to measure the propagation time over such a link.

NOTE—A PTP Link between PTP Ports of PTP Instances is also a gPTP Communication Path (see 3.9).

3.19 PTP Relay Instance: A PTP Instance that is capable of communicating synchronized time received on one port to other ports, using the IEEE 802.1AS protocol.

NOTE—A PTP Relay Instance could, for example, be contained in a bridge, a router, or a multi-port end station.

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

7.2.2 Time-aware network consisting of a single gPTP domain

Figure 7-1 illustrates an example time-aware network consisting of a single gPTP domain, using all theabove network technologies (i.e., (c) - (f) of 7.2.1), where end stations on several local networks are connected to a grandmaster on a backbone network via an EPON access network.

Any PTP Instance with clock sourcing capabilities can be a potential grandmaster, so there is a selection method (the *best master clock algorithm*, or BMCA) that ensures that all of the PTP Instances in a gPTP domain use the same grandmaster.[12] The BMCA is largely identical to that used in IEEE Std 1588-2019,

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10.3.1.1 Best master clock algorithm overview

In the BMCA (i.e., method (a) of 10.3.1), best master selection information is exchanged between PTP Instances of time-aware systems via Announce messages (see 10.5 and 10.6). Each Announce message contains time-synchronization spanning tree vector information that identifies one PTP Instance as the root of the time-synchronization spanning tree and, if the PTP Instance is grandmaster-capable, the grandmaster. Each PTP Instance in turn uses the information contained in the Announce messages it receives, along with its knowledge of itself, to compute which of the PTP Instances that it has knowledge of ought to be the root of the spanning tree and, if grandmaster-capable, the grandmaster. As part of constructing the time-synchronization spanning tree, each port of each PTP Instance is assigned a port state from Table 10-2 by state machines associated with the ports and with the PTP Instance as a whole.

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c) In gPTP there are only two types of PTP Instances: PTP End Instances and PTP Relay Instances, while IEEE 1588 has ordinary clocks, boundary clocks, end-to-end transparent clocks, and P2P transparent clocks. A PTP End Instance corresponds to an IEEE 1588 ordinary clock, and a PTP Relay Instance is a type of IEEE 1588 boundary clock where its operation is very tightly defined, so much so that a PTP Relay Instance with Ethernet ports can be shown to be mathematically equivalent to a P2P transparent clock in terms of how synchronization is performed, as shown in

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

IEEE Std 802.1Q-2018
IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks

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IEEE Std 802.1AS™, IEEE Standard for Local and metropolitan area networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.

(Source: https://avnu.org/wp-content/uploads/2014/05/as-kbstanton-8021AS-tutorial-0714-v01.pdf)

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IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks

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IEEE Std 802.1AS™, IEEE Standard for Local and metropolitan area networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

8.6.9.4.1 AdminBaseTime

The administrative value of base time, expressed as an IEEE 1588 precision time protocol (PTP) [B24] timescale (see 8.2 of IEEE Std 802.1AS™-2011 [B11]). This value can be changed by management, and is used by the List Config state machine (8.6.9.3) to set the value of OperBaseTime (8.6.9.4.18).

NOTE—Time is expressed in the PTP timescale as the number of seconds, nanoseconds, and fractional nanoseconds that have elapsed since 1 January 1970 00:00:00 TAI.

NOTE 2—If AdminBaseTime is set to the same time in the past in all bridges and end stations, OperBaseTime is always in the past, and all cycles start synchronized. Using AdminBaseTime in the past is appropriate when you can start schedules prior to starting the application that uses the schedules. Use of AdminBaseTime in the future is intended to change a currently running schedule in all bridges and end stations to a new schedule at a future time. Using AdminBaseTime in the future is appropriate when schedules must be changed without stopping the application.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

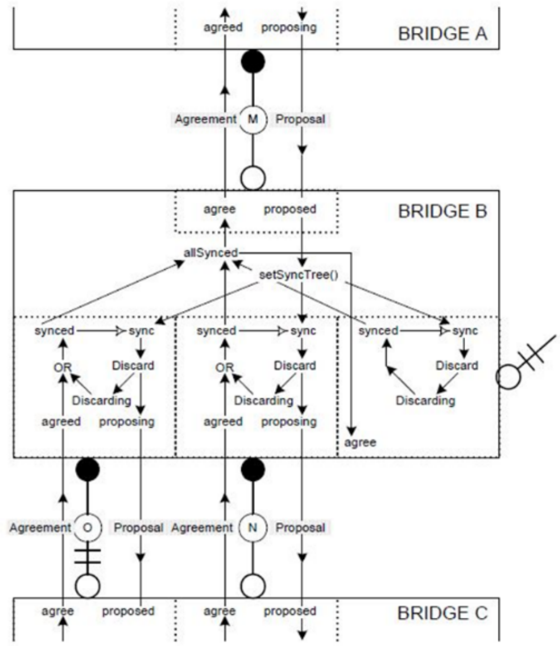

Figure 13-10—Agreements and Proposals

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

> NOTE 2—Agreements can be generated without prior receipt of a Proposal as soon as the necessary conditions are met. Subsequent receipt of a Proposal serves to elicit a further Agreement. If all other ports have already been synchronized (allSynced in Figure 13-10) and the Proposal's priority vector does not convey worse information, synchronization is maintained and there is no need to transition Designated Ports to Discarding once more, or to transmit further Proposals.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

65. The methods practiced by Ribbon's use of the accused products include maintaining a substantially synchronized time frame among the plurality of device adapters interconnected by a network, the time frame repeating periodically and including a plurality of

assigned time phases and a free access phase. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The time aware bridged LAN uses 802.1AS base time to maintain synchronized time for all the ports in the bridges (device adapters). The bridges containing ports schedule the transmission of traffic based on the synchronized time. Each port associated with a specific set of transmission queues includes a plurality of transmission gates. A transmission gate can be in a closed state or open state. The functionality of assigned time phases is achieved using open gates transmitting data packets during transmission time. The functionality of free access phase is achieved when the gates are opened for transmission during any time. IEEE Std. 802.1Q-2018 supports cyclic queuing and forwarding structures to create synchronized frames and gates which repeat periodically (Annex T).

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

5.2.b. Scope of the project: This amendment specifies time-aware queue-draining procedures, managed objects and extensions to existing protocols that enable bridges and end stations to schedule the transmission of frames based on a synchronized time. Virtual Local Area Network (VLAN) tag encoded priority values are allocated allowing simultaneous support of scheduled traffic, credit-based shaper traffic and other bridged traffic over Local Area Networks (LANs).

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

IEEE Std 802.1Q-2018
IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks

2. Normative references

The following referenced documents are indispensable for the application of this document (i.e., they must be understood and used, so each referenced document is cited in the text and its relationship to this document is explained). For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies.

ANSI X3.159, American National Standards for Information Systems—Programming Language—C.[5]

IEEE Std 802®, IEEE Standard for Local and metropolitan area networks: Overview and Architecture.[6,7]

IEEE Std 802.1AB™, IEEE Standard for Local and metropolitan area networks—Station and Media Access Control Connectivity Discovery.

IEEE Std 802.1AC™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Service Definition.

IEEE Std 802.1AE™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Security.

IEEE Std 802.1AS™, IEEE Standard for Local and metropolitan area networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.

(Source: https://avnu.org/wp-content/uploads/2014/05/as-kbstanton-8021AS-tutorial-0714-v01.pdf)

3.16 PTP End Instance: A PTP Instance that is capable of acting as the source of synchronized time on the network, or destination of synchronized time using the IEEE 802.1AS protocol, or both.

3.17 PTP Instance: A PTP Relay Instance or a PTP End Instance. A PTP Instance implements those portions of this standard indicated as applicable to a PTP Relay Instance or a PTP End Instance. Each PTP Instance operates in exactly one domain.

3.18 PTP Link: Within a domain, a network segment between two PTP Ports using the peer-to-peer delay mechanism of this standard. The peer-to-peer delay mechanism is designed to measure the propagation time over such a link.

NOTE—A PTP Link between PTP Ports of PTP Instances is also a gPTP Communication Path (see 3.9).

3.19 PTP Relay Instance: A PTP Instance that is capable of communicating synchronized time received on one port to other ports, using the IEEE 802.1AS protocol.

NOTE—A PTP Relay Instance could, for example, be contained in a bridge, a router, or a multi-port end station.

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

8.6.8.4 Enhancements for scheduled traffic

A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:

a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

8.6.8 Transmission selection

For each Port, frames are selected for transmission on the basis of the traffic classes that the Port supports and the operation of the transmission selection algorithms supported by the corresponding queues on that Port. For a given Port and traffic class, frames are selected from the corresponding queue for transmission if and only if

a) The operation of the transmission selection algorithm supported by that queue determines that there is a frame available for transmission; and

b) For each queue corresponding to a numerically higher value of traffic class supported by the Port, the operation of the transmission selection algorithm supported by that queue determines that there is no frame available for transmission.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

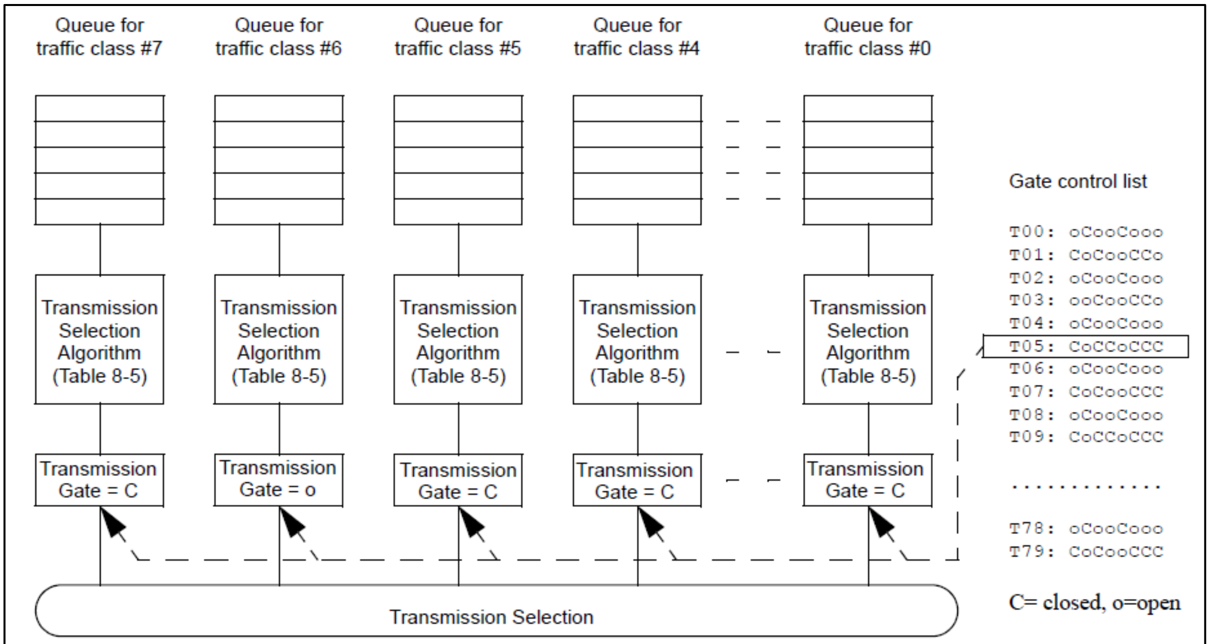

Figure 8-12—Transmission selection with gates

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

> A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

> **3.3 gating cycle:** The period of time over which the sequence of operations in a gate control list repeats.
>
> **3.4 transmission gate:** A gate that connects or disconnects the transmission selection function of the forwarding process from the queue, allowing or preventing it from selecting frames from that queue. The gate has two states, Open and Closed.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

Cyclic queuing and forwarding[53]

T.1 Overview of CQF

Cyclic queuing and forwarding (CQF) is a method of traffic shaping that can deliver deterministic, and easily calculated, latency for time-sensitive traffic streams. As the name implies, the principle underlying CQF is that stream traffic is transmitted and queued for transmission along a network path in a cyclic manner. Time is divided into numbered time intervals i, i+1, i+2, ... i+N, each of duration d. Frames transmitted by a Bridge, *Alice*, during time interval i are received by a downstream Bridge, *Bob*, during time interval i and are transmitted onwards by *Bob* towards Bridge *Charlie* during time interval i+1, and so on. A starting assumption is that, for a given traffic class, all Bridges and all end stations connected to a given bridge have a common understanding (to a known accuracy) of the start time of cycle i, and the cycle duration, d.

Frames transmitted by *Alice* during interval i are transmitted by *Bob* in interval i+1; the maximum possible delay experienced by a given frame is from the beginning of i to the end of i+1, or twice d. Similarly, the minimum possible delay experienced is from the end of i to the beginning of i+1, which is zero. More generally, the maximum delay experienced by a given frame is

$$(h+1) \times d$$

and the minimum delay experienced by a given frame is

$$(h-1) \times d$$

where h is the number of hops.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

66. The methods practiced by Ribbon's use of the accused products include assigning each time phase to a specific device adapter prior to transmission of real-time data by the specific device adapter. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. IEEE standard 802.1Q shows that scheduling (assigning) of transmission gates starts prior to the transmission of real-time data. Per clause 8.6.8 of the IEEE Std. 802.1Q, each time phase is assigned to a specific device adapter prior to transmission of real-time data by the specific device adapter. IEEE 802.1Q performs traffic shaping through Per-Stream Filtering and Policing (PSTP).

NOTE 2—If AdminBaseTime is set to the same time in the past in all bridges and end stations, OperBaseTime is always in the past, and all cycles start synchronized. Using AdminBaseTime in the past is appropriate when you can start schedules prior to starting the application that uses the schedules. Use of AdminBaseTime in the future is intended to

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

5.5 Need for the Project: The credit-based shaper works well in arbitrary networks (i.e., non-engineered). Networks employing scheduled transmissions are able to control real-time processes. This amendment enables those two kinds of networks to be consolidated into a single network, with a significant cost reduction to the user.

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

8.6.8 Transmission selection

For each Port, frames are selected for transmission on the basis of the traffic classes that the Port supports and the operation of the transmission selection algorithms supported by the corresponding queues on that Port. For a given Port and traffic class, frames are selected from the corresponding queue for transmission if and only if

a) The operation of the transmission selection algorithm supported by that queue determines that there is a frame available for transmission; and

b) For each queue corresponding to a numerically higher value of traffic class supported by the Port, the operation of the transmission selection algorithm supported by that queue determines that there is no frame available for transmission.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

5.4.1.9 Cyclic queuing and forwarding (CQF) requirements

A VLAN Bridge component implementation that conforms to the provisions of this standard for CQF (see Annex T) shall

a) Support the enhancements for scheduled traffic as specified in 8.6.8.4.

b) Support the state machines for scheduled traffic as specified in 8.6.9.
c) Support the state machines for stream gate control as specified in 8.6.10.
d) Support the management entities for scheduled traffic as specified in 12.29.
e) Support the requirements for per-stream filtering and policing (PSFP) as stated in 5.4.1.8.
f) Support the management entities for PSFP as specified in 12.31.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

5.4.1.5 Forwarding and Queuing Enhancements for time-sensitive streams (FQTSS)—requirements

A VLAN Bridge component implementation that conforms to the provisions of this standard for FQTSS shall

a) Support a minimum of two traffic classes on all Ports, of which
   1) A minimum of one traffic class supports the strict priority algorithm for transmission selection (8.6.8.1), and
   2) One traffic class is a stream reservation (SR) class.
b) Support the operation of the credit-based shaper algorithm (8.6.8.2) on all Ports as the transmission selection algorithm used for the SR class.
c) Support SRP domain boundary port priority regeneration override as defined in 6.9.4, and the default priority regeneration override value defined in Table 6-5, for SR class "B."
d) Support the tables and procedures for mapping priorities to traffic classes as defined in 34.5.

SR: Stream Reservation, SRP: Stream Reservation Protocol

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

67. The plurality of device adaptors recited above in connection with Ribbon's use of the accused products includes a respective one of the plurality of device adapters that is configured to transmit data during at least one of a respective assigned time phase or the free access phase, to refrain from transmitting data during time phases not assigned to the respective one of the plurality of device adapters, and to determine whether to defer transmission of data during at least one of the assigned time phase or the free access phase to allow a non-real time device to transmit data. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. IEEE standard 802.1Q implements a method in which open gates transmit data packets during transmission time and closed gates refrain data packets from transmission. The functionality of assigned time phases is achieved using open gates transmitting data packets during transmission time. The functionality of free access phase is achieved by the gates that are opened for transmission during any time. IEEE Std. 802.1Q supports Forwarding and Queuing Enhancements for Time Sensitive Streams. Thus, one of the plurality of device adaptors is configured to transmit data during at least one of a respective assigned time phase or

free access phase, to refrain from transmitting data during time phases not assigned to the respective one of the plurality of device adapters. IEEE Std. 802.1Q provides traffic shaping for various classes of data transmission and determining whether to defer transmission of data during at least one of the assigned time phase or the free access phase to allow a non-real time device to transmit data.

8.6.8.4 Enhancements for scheduled traffic

A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:

a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

5.4.1.5 Forwarding and Queuing Enhancements for time-sensitive streams (FQTSS)—requirements

A VLAN Bridge component implementation that conforms to the provisions of this standard for FQTSS shall

a) Support a minimum of two traffic classes on all Ports, of which
   1) A minimum of one traffic class supports the strict priority algorithm for transmission selection (8.6.8.1), and
   2) One traffic class is a stream reservation (SR) class.

b) Support the operation of the credit-based shaper algorithm (8.6.8.2) on all Ports as the transmission selection algorithm used for the SR class.

c) Support SRP domain boundary port priority regeneration override as defined in 6.9.4, and the default priority regeneration override value defined in Table 6-5, for SR class "B."

d) Support the tables and procedures for mapping priorities to traffic classes as defined in 34.5.

SR: Stream Reservation, SRP: Stream Reservation Protocol

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

5.4.1.6 ETS Bridge requirements

A device supporting ETS shall

a) Support at least 3 traffic classes (37.3).

NOTE—A minimum of 3 traffic classes allows a minimum configuration such that one traffic class contains priorities with PFC enabled, one traffic class contains priorities with PFC disabled, and one traffic class using strict priority.

b) Support bandwidth configuration with a granularity of 1% or finer (37.3).

c) Support bandwidth allocation with a precision of 10% (37.3).

d) Support a transmission selection policy such that if one of the traffic classes does not consume its allocated bandwidth, then any unused bandwidth is available to other traffic classes (37.3).

e) Support DCBX (Clause 38).

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

68. Ribbon has had knowledge of the '053 Patent at least as of the date when it was notified of the filing of this action.

69. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

70. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the ‘053 Patent.

COUNT V

DIRECT INFRINGEMENT OF U.S. PATENT NO. 6,246,702

71. On June 12, 2001, United States Patent No. 6,246,702 (“the ‘702 Patent”) was duly and legally issued by the United States Patent and Trademark Office for an invention entitled “Methods and Apparatus for Providing Quality-of-Service Guarantees in Computer Networks.”

72. Far North Patents is the owner of the ‘702 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the ‘702 Patent against infringers, and to collect damages for all relevant times.

73. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Ribbon/Edgewater EdgeMarc 2900 enterprise session border controller family of products that include advanced quality of service capabilities (collectively, “accused products”).

edgewater NETWORKS

EdgeMarc 2900e and 2900a POE

INTELLIGENT EDGE - ENTERPRISE SESSION BORDER CONTROLLERS

VLANs	• 802.1Q (up to 16) • Multi-VLAN ALG support

(Source : https://www.teledynamics.com/tdresources/d82d2f8c-7cbd-44dc-aa37-68d5b81a3ef8.pdf)

Edgewater Networks is Now Ribbon

The full suite of the Edgewater solutions and products are available from Ribbon.

(Source : https://info.rbbn.com/edgewater/)

Edgewater Networks Announces Launch of the EdgeMarc 2900 Series

June 06, 2017 12:00 PM Eastern Daylight Time

SAN JOSE, Calif.--(BUSINESS WIRE)--Edgewater Networks, the market leader in Network Edge Orchestration, announced the availability of the EdgeMarc 2900 Intelligent Edge devices. The EdgeMarc 2900 series enables enterprises and service providers to future-proof their SIP trunking and Unified Communications deployments, providing a highly flexible, scalable, and secure platform for service delivery and ongoing service quality management.

EdgeMarc 2900 Intelligent Edge breaks new ground in #NetworkEdgeOrchestration to future-proof SIP trunking and UC.

Tweet this

The EdgeMarc 2900e offers dual Ethernet and Optical WAN connections to extend the capabilities of Edgewater Networks' Network Edge Orchestration platform and align with Edgewater Networks' SD-WAN solution. The EdgeMarc 2900e supports WAN connections up to 1 Gbps and can be used for Small and Medium Business applications or Mid-Market deployments up to 300 concurrent calls. "We're excited to announce the availability of the first Intelligent Edge in our EdgeMarc 2900 series that enables service providers to deliver optimized VoIP solutions to nearly every use case in the market," said Chris Kolstad, VP of Product Management at Edgewater Networks. "The inherent flexibility and power of the EdgeMarc 2900 enables deployments in SMB environments with as few as five seats to enterprise environments with thousands of seats."

Future Intelligent Edge products will include the EdgeMarc 2900a and the EdgeMarc 2900POE. The EdgeMarc 2900a will add both outbound and inbound analog line (FXO/FXS) support for applications such as PSTN fail-over, fax, and overhead paging. The EdgeMarc 2900POE will integrate managed POE ports to provide an all-in-one solution for small office implementations.

(Source : https://www.businesswire.com/news/home/20170606005340/en/Edgewater-Networks-Announces-Launch-EdgeMarc-2900-Series)

74. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 27 of the '702 Patent. Ribbon's infringement in this regard is ongoing.

75. Ribbon has infringed the '702 Patent by using the accused products and thereby practicing a method for regulating traffic in an Ethernet network including real-time devices, non-real-time devices, a network medium, and a plurality of device adapters connected between the devices and the network medium, each of the device adapters including a clock. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. IEEE standard 802.1Q (Qbv) implements a method for scheduling traffic in time aware Local Area Network (Ethernet network). The time aware network includes multiple end points

(devices) and bridges that includes ports (device adapters). The endpoint and bridges are connected via interconnects (network medium). Also, each time aware bridge including port is a boundary clock. The endpoints transmit data that is a mix of time-critical traffic and other traffic, i.e, real time data and non-real time data. The end point is either a real time device or a non-real time device. IEEE Std. 802.1AS™ is normative and essential to implement an IEEE Std. 802.1Q Compliant System.

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

5.2.b. Scope of the project: This amendment specifies time-aware queue-draining procedures, managed objects and extensions to existing protocols that enable bridges and end stations to schedule the transmission of frames based on a synchronized time. Virtual Local Area Network (VLAN) tag encoded priority values are allocated allowing simultaneous support of scheduled traffic, credit-based shaper traffic and other bridged traffic over Local Area Networks (LANs).

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

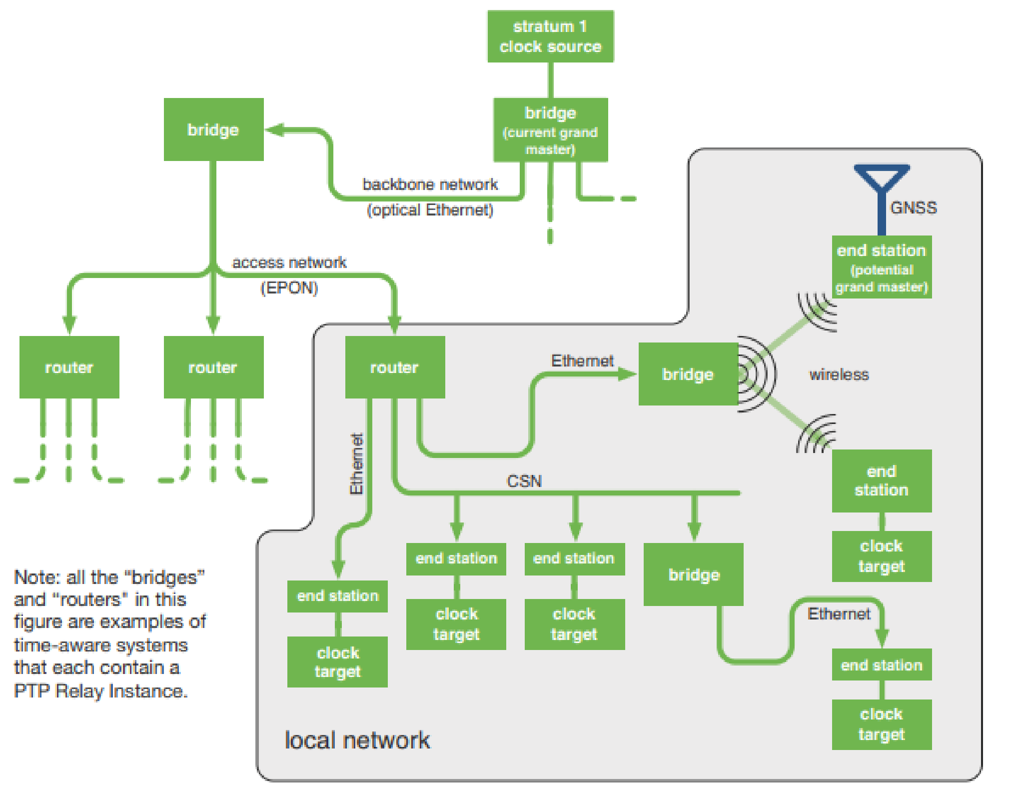

Figure 7-1—Time-aware network example

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

7.2 Architecture of a time-aware network

7.2.1 General

A time-aware network consists of a number of interconnected time-aware systems that support the gPTP defined within this standard. These time-aware systems can be any networking device, including, for example, bridges, routers, and end stations. A set of time-aware systems that are interconnected by gPTP-capable network elements is called a *gPTP network*. Each instance of gPTP that the time-aware systems support is in one *gPTP domain*, and the instances of gPTP are said to be part of that gPTP domain. A time-aware system can support, and therefore be part of, more than one gPTP domain. The entity of a single time-aware system that executes gPTP in one gPTP domain is called a PTP Instance. A time-aware system can contain multiple PTP Instances, which are each associated with a different gPTP domain. There are two

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

c) In gPTP there are only two types of PTP Instances: PTP End Instances and PTP Relay Instances, while IEEE 1588 has ordinary clocks, boundary clocks, end-to-end transparent clocks, and P2P transparent clocks. A PTP End Instance corresponds to an IEEE 1588 ordinary clock, and a PTP Relay Instance is a type of IEEE 1588 boundary clock where its operation is very tightly defined, so much so that a PTP Relay Instance with Ethernet ports can be shown to be mathematically equivalent to a P2P transparent clock in terms of how synchronization is performed, as shown in

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

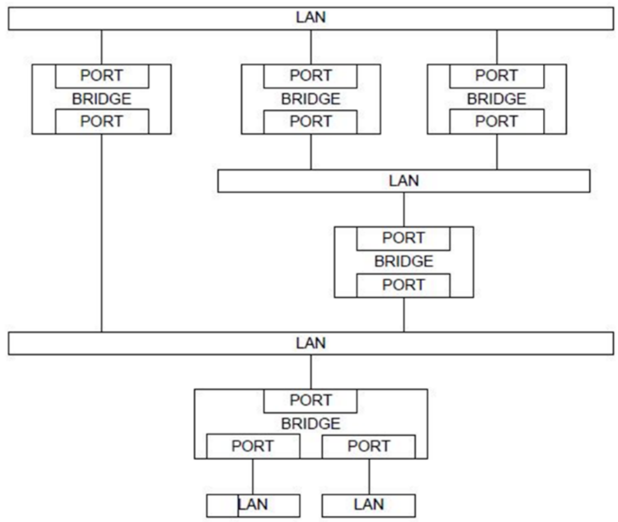

Figure 8-1—A Bridged Network

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

Traffic scheduling

This annex provides some background to the mechanisms provided in this standard for traffic scheduling, with the intent of providing some insight into the motivation for the provision of mechanisms for traffic scheduling and some of the ways that the mechanisms might be used.

Q.1 Motivation

Some applications have a need for frame delivery that is highly predictable in terms of the time at which frame transmission will occur, and the overall latency and jitter that will be experienced as the frame is propagated to its destination. Examples include industrial and automotive control applications, where data transmitted over the network is used to feed the parameters of control loops that are critical to the operation of the plant or machinery involved, and where frames carrying control data are transmitted on a repeating time schedule; late delivery of such frames can result in instability, inaccuracy, or failure of the operation of the control loops concerned. In some implementations, this need has been met by the provision of dedicated, highly engineered networks that are used solely for the transmission of time-critical control traffic; however, as the bandwidth occupied by such traffic is often low, and the cost of providing a dedicated control network can be high, it can be desirable to mix time-critical traffic with other classes of traffic in the same network, as long as such mixing can be achieved while still meeting the timing requirements of the time-critical traffic.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

IEEE Std 802.1Q-2018
IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks

2. Normative references

The following referenced documents are indispensable for the application of this document (i.e., they must be understood and used, so each referenced document is cited in the text and its relationship to this document is explained). For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies.

ANSI X3.159, American National Standards for Information Systems—Programming Language—C.[5]

IEEE Std 802®, IEEE Standard for Local and metropolitan area networks: Overview and Architecture.[6,7]

IEEE Std 802.1AB™, IEEE Standard for Local and metropolitan area networks—Station and Media Access Control Connectivity Discovery.

IEEE Std 802.1AC™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Service Definition.

IEEE Std 802.1AE™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Security.

IEEE Std 802.1AS™, IEEE Standard for Local and metropolitan area networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

76. The methods practiced by Ribbon's use of the accused products include defining a common time reference for the device adapters, said common time reference including a frame of time having a plurality of time phases, each of device adapters being uniquely assigned to one of said plurality of time phases, said plurality of time phases including a free-access phase. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The time aware bridged LAN use 802.1AS base time to maintain synchronized time (common time reference) for all the ports in the bridges (device adapters). The bridges containing ports schedule the transmission of traffic based on the synchronized time. Each port associated with a specific set of transmission queues includes a plurality of transmission gates. A transmission gate can be in a closed state or open state. The functionality of assigned time phases is achieved using open

gates transmitting data packets during transmission time. The functionality of free access phase is achieved when the gates are opened for transmission during any time.

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

5.2.b. Scope of the project: This amendment specifies time-aware queue-draining procedures, managed objects and extensions to existing protocols that enable bridges and end stations to schedule the transmission of frames based on a synchronized time. Virtual Local Area Network (VLAN) tag encoded priority values are allocated allowing simultaneous support of scheduled traffic, credit-based shaper traffic and other bridged traffic over Local Area Networks (LANs).

(Source: http://www.ieee802.org/1/files/public/docs2014/bv-p802-1Qbv-par-modification-1114.pdf)

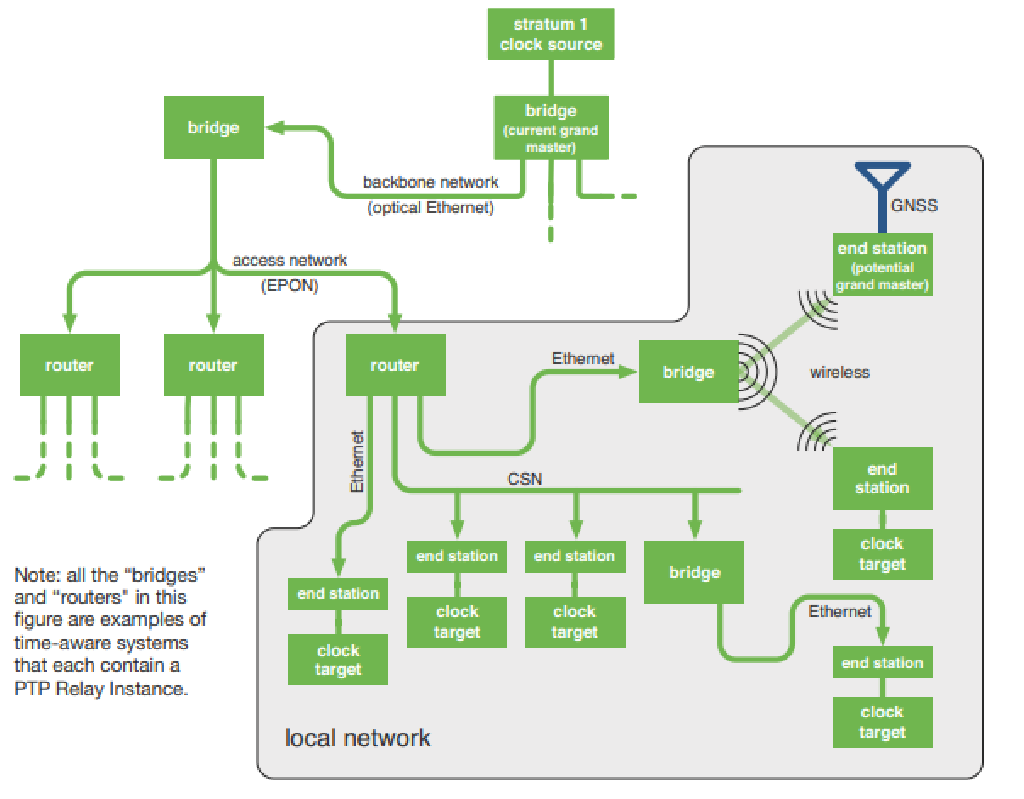

Figure 7-1—Time-aware network example

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

3.16 PTP End Instance: A PTP Instance that is capable of acting as the source of synchronized time on the network, or destination of synchronized time using the IEEE 802.1AS protocol, or both.

3.17 PTP Instance: A PTP Relay Instance or a PTP End Instance. A PTP Instance implements those portions of this standard indicated as applicable to a PTP Relay Instance or a PTP End Instance. Each PTP Instance operates in exactly one domain.

3.18 PTP Link: Within a domain, a network segment between two PTP Ports using the peer-to-peer delay mechanism of this standard. The peer-to-peer delay mechanism is designed to measure the propagation time over such a link.

NOTE—A PTP Link between PTP Ports of PTP Instances is also a gPTP Communication Path (see 3.9).

3.19 PTP Relay Instance: A PTP Instance that is capable of communicating synchronized time received on one port to other ports, using the IEEE 802.1AS protocol.

NOTE—A PTP Relay Instance could, for example, be contained in a bridge, a router, or a multi-port end station.

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

8.6.8.4 Enhancements for scheduled traffic

A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:

a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

8.6.8 Transmission selection

For each Port, frames are selected for transmission on the basis of the traffic classes that the Port supports and the operation of the transmission selection algorithms supported by the corresponding queues on that Port. For a given Port and traffic class, frames are selected from the corresponding queue for transmission if and only if

a) The operation of the transmission selection algorithm supported by that queue determines that there is a frame available for transmission; and

b) For each queue corresponding to a numerically higher value of traffic class supported by the Port, the operation of the transmission selection algorithm supported by that queue determines that there is no frame available for transmission.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

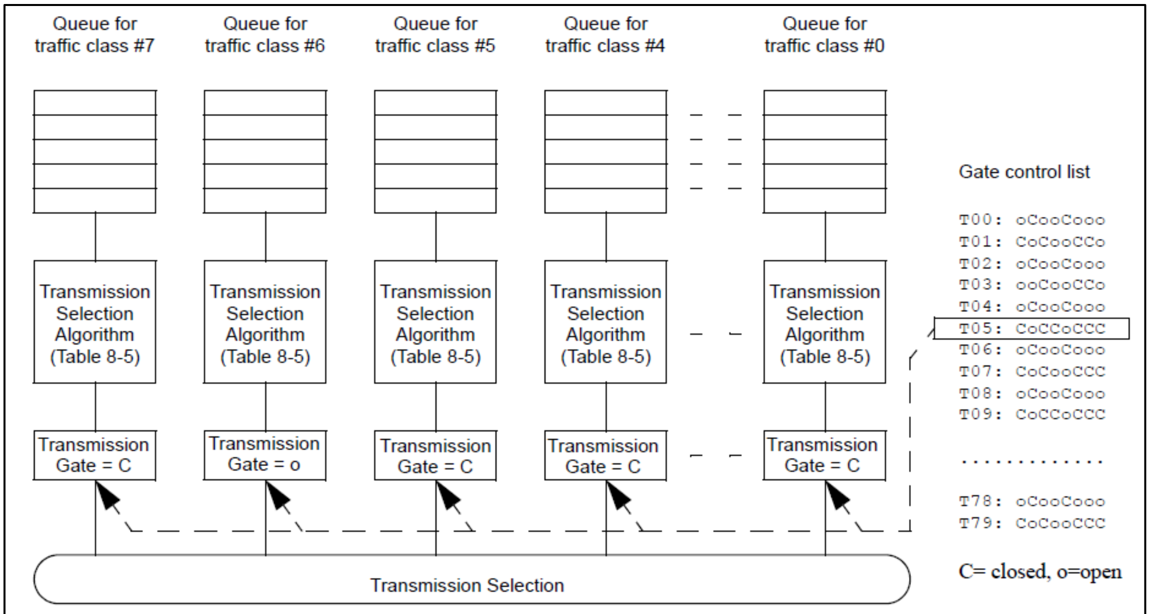

Figure 8-12—Transmission selection with gates

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

> A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

77. The methods practiced by Ribbon's use of the accused products include allowing a device adapter to transmit packets during said time phase uniquely assigned thereto and during said free-access phase. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The open gates assigned to the transmission queue of a port (device adapter) transmit the data packets during the transmission time. The transmission selection algorithm determines the transmission time. If the gates are in permanent open state, the data packets are transmitted from the gates during any time. The functionality of assigned time phases

is achieved using open gates transmitting data packets during transmission time. The functionality of free access phase is achieved when the gates are permanently opened for transmission during any time.

8.6.8.4 Enhancements for scheduled traffic

A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:

a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

3.4 transmission gate: A gate that connects or disconnects the transmission selection function of the forwarding process from the queue, allowing or preventing it from selecting frames from that queue. The gate has two states, Open and Closed.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

78. The methods practiced by Ribbon's use of the accused products include designating one of said device adapters as a master timing device. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The ports of the time

aware bridge (device adapter) uses best master clock algorithm (BMCA) to determine a potential grandmaster port (master timing device).

7.2.2 Time-aware network consisting of a single gPTP domain

Figure 7-1 illustrates an example time-aware network consisting of a single gPTP domain, using all theabove network technologies (i.e., (c) - (f) of 7.2.1), where end stations on several local networks are connected to a grandmaster on a backbone network via an EPON access network.

Any PTP Instance with clock sourcing capabilities can be a potential grandmaster, so there is a selection method (the *best master clock algorithm*, or BMCA) that ensures that all of the PTP Instances in a gPTP domain use the same grandmaster.[12] The BMCA is largely identical to that used in IEEE Std 1588-2019,

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

10.3.1.1 Best master clock algorithm overview

In the BMCA (i.e., method (a) of 10.3.1), best master selection information is exchanged between PTP Instances of time-aware systems via Announce messages (see 10.5 and 10.6). Each Announce message contains time-synchronization spanning tree vector information that identifies one PTP Instance as the root of the time-synchronization spanning tree and, if the PTP Instance is grandmaster-capable, the grandmaster. Each PTP Instance in turn uses the information contained in the Announce messages it receives, along with its knowledge of itself, to compute which of the PTP Instances that it has knowledge of ought to be the root of the spanning tree and, if grandmaster-capable, the grandmaster. As part of constructing the time-synchronization spanning tree, each port of each PTP Instance is assigned a port state from Table 10-2 by state machines associated with the ports and with the PTP Instance as a whole.

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

11.1.3 Transport of time-synchronization information

The transport of time-synchronization information by a PTP Instance, using Sync and Follow_Up (or just Sync) messages, is illustrated in Figure 11-2. The mechanism is mathematically equivalent to the mechanism described in IEEE Std 1588-2019 for a peer-to-peer transparent clock that is syntonized (see 11.4.5.1, 11.5.1, and 11.5.2.2 of IEEE Std 1588-2019). However, as will be seen shortly, the processes of transporting synchronization by a peer-to-peer transparent clock that is syntonized and by a boundary clock are mathematically and functionally equivalent. The main functional difference between the two types of clocks is that the boundary clock participates in best master selection and invokes the BMCA, while the peer-to-peer transparent clock does not participate in best master selection and does not invoke the BMCA (and implementations of the two types of clocks can be different).

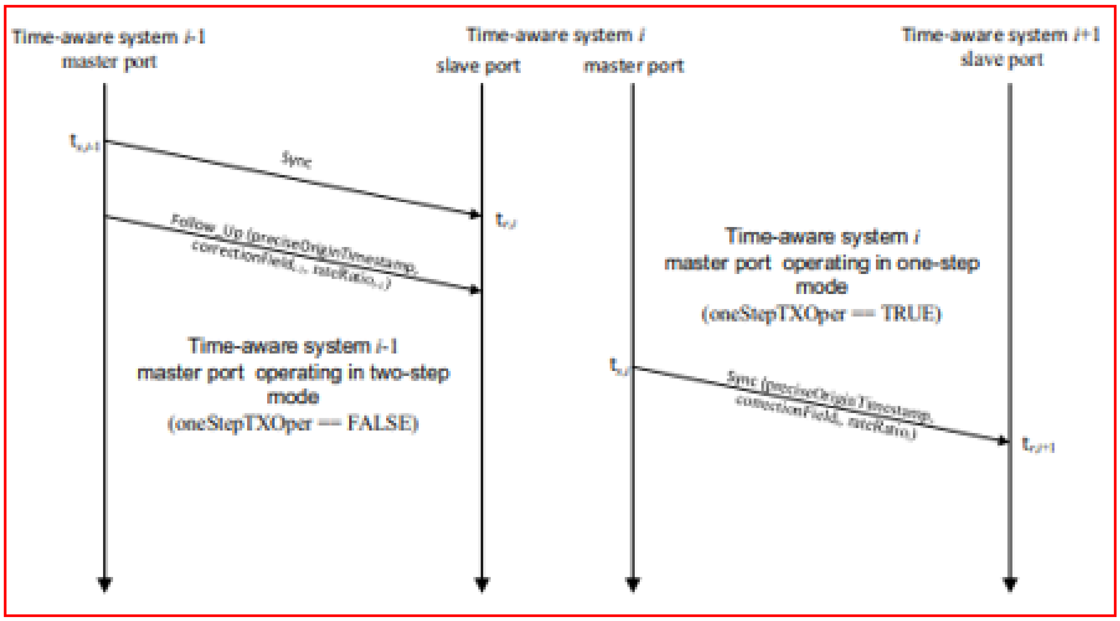

Figure 11-2—Transport of time-synchronization information

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

c) In gPTP there are only two types of PTP Instances: PTP End Instances and PTP Relay Instances, while IEEE 1588 has ordinary clocks, boundary clocks, end-to-end transparent clocks, and P2P transparent clocks. A PTP End Instance corresponds to an IEEE 1588 ordinary clock, and a PTP Relay Instance is a type of IEEE 1588 boundary clock where its operation is very tightly defined, so much so that a PTP Relay Instance with Ethernet ports can be shown to be mathematically equivalent to a P2P transparent clock in terms of how synchronization is performed, as shown in

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

79. The methods practiced by Ribbon's use of the accused products include synchronizing the clocks of the remaining device adapters with said master timing device. For

example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The boundary clocks of the slave ports in the time aware bridges (remaining device adapters) are synchronized with the grandmaster boundary clock (master timing device).

> c) In gPTP there are only two types of PTP Instances: PTP End Instances and PTP Relay Instances, while IEEE 1588 has ordinary clocks, boundary clocks, end-to-end transparent clocks, and P2P transparent clocks. A PTP End Instance corresponds to an IEEE 1588 ordinary clock, and a PTP Relay Instance is a type of IEEE 1588 boundary clock where its operation is very tightly defined, so much so that a PTP Relay Instance with Ethernet ports can be shown to be mathematically equivalent to a P2P transparent clock in terms of how synchronization is performed, as shown in

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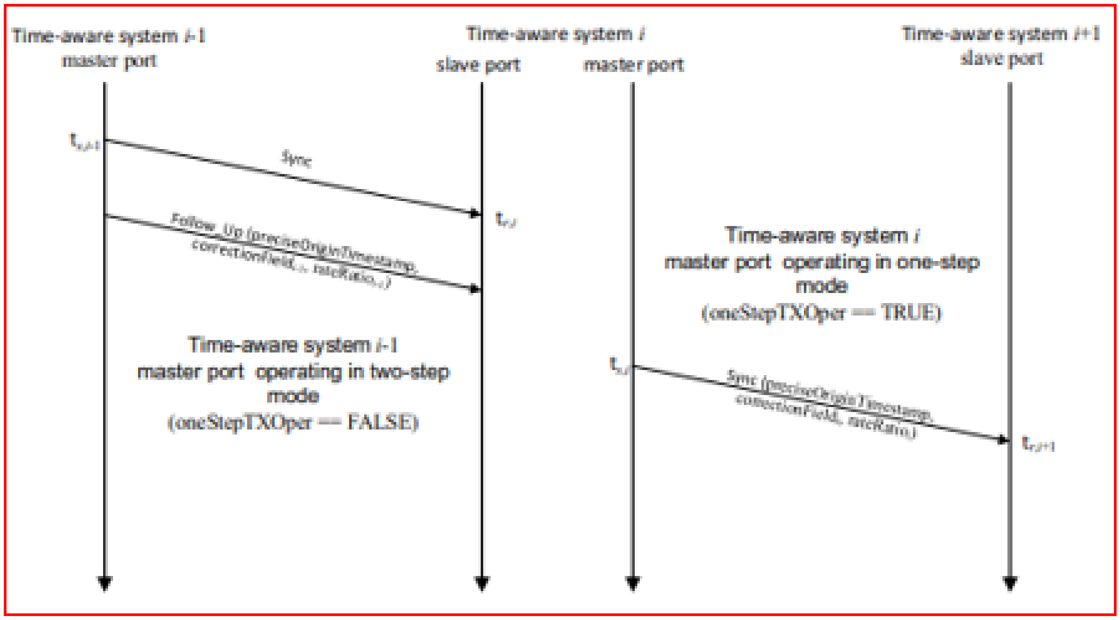


Figure 11-2—Transport of time-synchronization information

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

80. Ribbon has had knowledge of the '702 Patent at least as of the date when it was notified of the filing of this action.

81. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

82. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the '702 Patent.

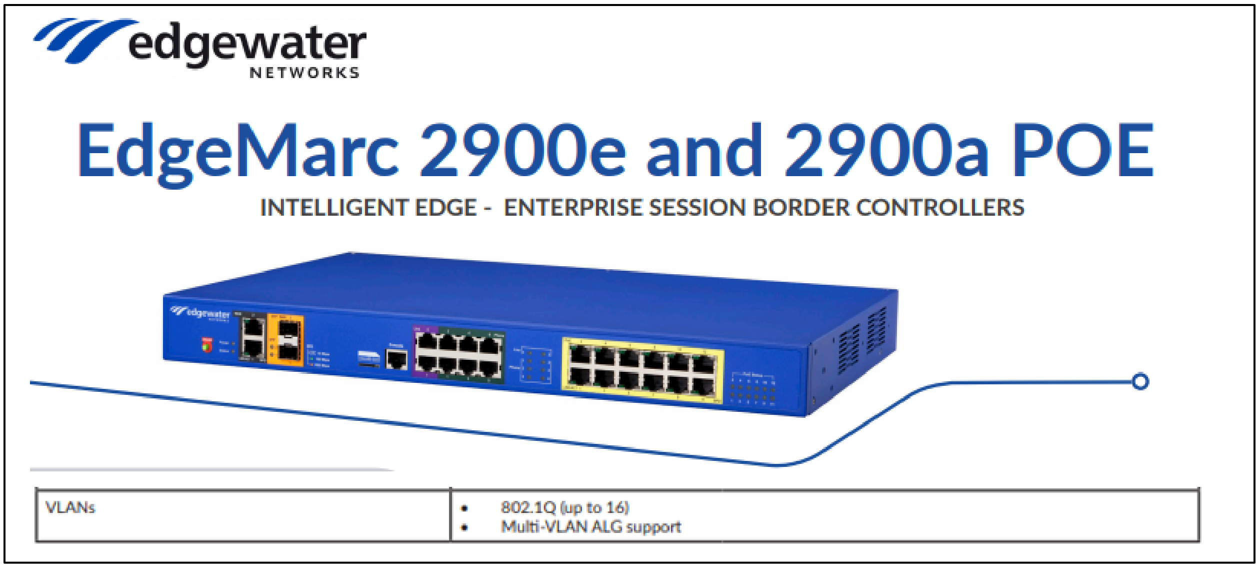
COUNT VI

DIRECT INFRINGEMENT OF U.S. PATENT NO. 6,215,797

83. On April 10, 2001, United States Patent No. 6,215,797 ("the '797 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Methods and Apparatus for Providing Quality of Service Guarantees in Computer Networks."

84. Far North Patents is the owner of the '797 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '797 Patent against infringers, and to collect damages for all relevant times.

85. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Ribbon/Edgewater EdgeMarc 2900 enterprise session border controller family of products that include advanced quality of service capabilities (collectively, "accused products").


edgewater NETWORKS

EdgeMarc 2900e and 2900a POE

INTELLIGENT EDGE - ENTERPRISE SESSION BORDER CONTROLLERS

VLANs	• 802.1Q (up to 16) • Multi-VLAN ALG support

(Source : https://www.teledynamics.com/tdresources/d82d2f8c-7cbd-44dc-aa37-68d5b81a3ef8.pdf)

(Source : https://info.rbbn.com/edgewater/)

Edgewater Networks Announces Launch of the EdgeMarc 2900 Series

June 06, 2017 12:00 PM Eastern Daylight Time

SAN JOSE, Calif.--(BUSINESS WIRE)--Edgewater Networks, the market leader in Network Edge Orchestration, announced the availability of the EdgeMarc 2900 Intelligent Edge devices. The EdgeMarc 2900 series enables enterprises and service providers to future-proof their SIP trunking and Unified Communications deployments, providing a highly flexible, scalable, and secure platform for service delivery and ongoing service quality management.

EdgeMarc 2900 Intelligent Edge breaks new ground in #NetworkEdgeOrchestration to future-proof SIP trunking and UC.

Tweet this

The EdgeMarc 2900e offers dual Ethernet and Optical WAN connections to extend the capabilities of Edgewater Networks' Network Edge Orchestration platform and align with Edgewater Networks' SD-WAN solution. The EdgeMarc 2900e supports WAN connections up to 1 Gbps and can be used for Small and Medium Business applications or Mid-Market deployments up to 300 concurrent calls. "We're excited to announce the availability of the first Intelligent Edge in our EdgeMarc 2900 series that enables service providers to deliver optimized VoIP solutions to nearly every use case in the market," said Chris Kolstad, VP of Product Management at Edgewater Networks. "The inherent flexibility and power of the EdgeMarc 2900 enables deployments in SMB environments with as few as five seats to enterprise environments with thousands of seats."

Future Intelligent Edge products will include the EdgeMarc 2900a and the EdgeMarc 2900POE. The EdgeMarc 2900a will add both outbound and inbound analog line (FXO/FXS) support for applications such as PSTN fail-over, fax, and overhead paging. The EdgeMarc 2900POE will integrate managed POE ports to provide an all-in-one solution for small office implementations.

(Source : https://www.businesswire.com/news/home/20170606005340/en/Edgewater-Networks-Announces-Launch-EdgeMarc-2900-Series)

86. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 30 of the '797 Patent. Ribbon's infringement in this regard is ongoing.

87. Ribbon has infringed the '797 Patent by using the accused products and thereby practicing a method for regulating traffic in a network including devices for generating packets of data, a network medium for carrying the packets, and a plurality of device adapters connected between the devices and the network medium. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. IEEE standard 802.1Q (Qbv) implements a method for scheduling traffic in time aware Local Area Network. The time aware network includes multiple end points that transmit data packets and bridges that includes ports (device

adapters). The interconnects (network medium) carry the transmitted data packets. The endpoints and bridges are connected via interconnects.

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

5.2.b. Scope of the project: This amendment specifies time-aware queue-draining procedures, managed objects and extensions to existing protocols that enable bridges and end stations to schedule the transmission of frames based on a synchronized time. Virtual Local Area Network (VLAN) tag encoded priority values are allocated allowing simultaneous support of scheduled traffic, credit-based shaper traffic and other bridged traffic over Local Area Networks (LANs).

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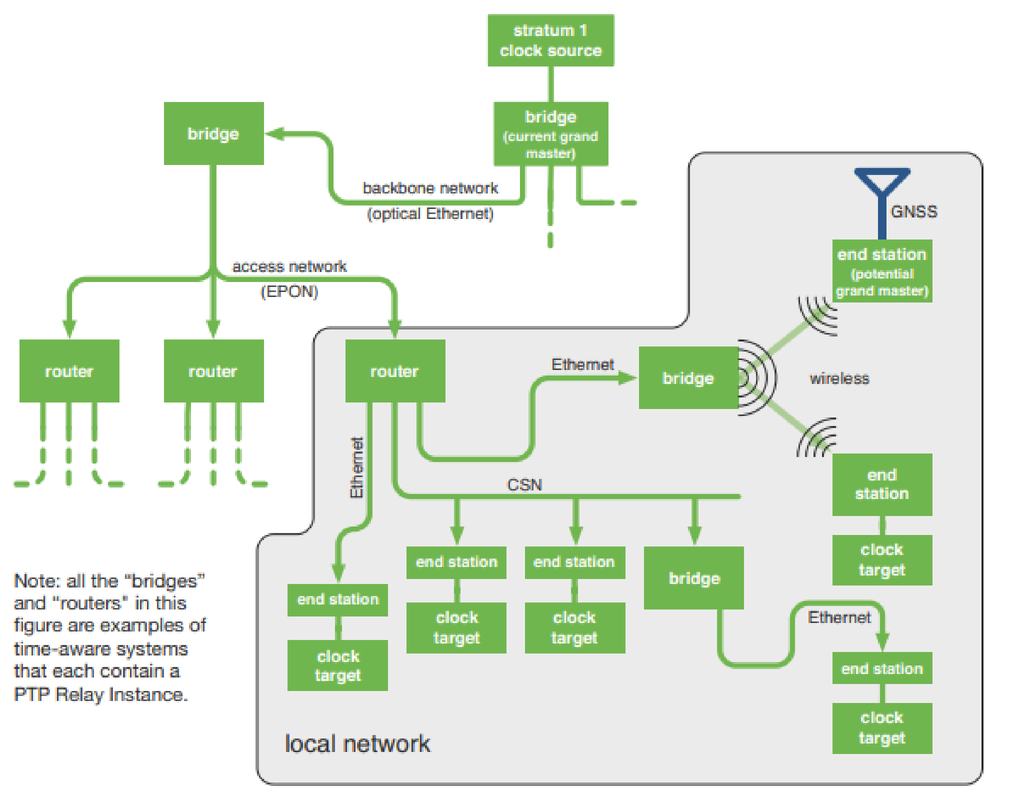

Figure 7-1—Time-aware network example

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

7.2 Architecture of a time-aware network

7.2.1 General

A time-aware network consists of a number of interconnected time-aware systems that support the gPTP defined within this standard. These time-aware systems can be any networking device, including, for example, bridges, routers, and end stations. A set of time-aware systems that are interconnected by gPTP-capable network elements is called a *gPTP network*. Each instance of gPTP that the time-aware systems support is in one *gPTP domain*, and the instances of gPTP are said to be part of that gPTP domain. A time-aware system can support, and therefore be part of, more than one gPTP domain. The entity of a single time-aware system that executes gPTP in one gPTP domain is called a PTP Instance. A time-aware system can contain multiple PTP Instances, which are each associated with a different gPTP domain. There are two

(Source: https://1.ieee802.org/wp-content/uploads/2019/03/802-1AS-rev-d8-0.pdf)

Traffic scheduling

This annex provides some background to the mechanisms provided in this standard for traffic scheduling, with the intent of providing some insight into the motivation for the provision of mechanisms for traffic scheduling and some of the ways that the mechanisms might be used.

Q.1 Motivation

Some applications have a need for frame delivery that is highly predictable in terms of the time at which frame transmission will occur, and the overall latency and jitter that will be experienced as the frame is propagated to its destination. Examples include industrial and automotive control applications, where data transmitted over the network is used to feed the parameters of control loops that are critical to the operation of the plant or machinery involved, and where frames carrying control data are transmitted on a repeating time schedule; late delivery of such frames can result in instability, inaccuracy, or failure of the operation of the control loops concerned. In some implementations, this need has been met by the provision of dedicated, highly engineered networks that are used solely for the transmission of time-critical control traffic; however, as the bandwidth occupied by such traffic is often low, and the cost of providing a dedicated control network can be high, it can be desirable to mix time-critical traffic with other classes of traffic in the same network, as long as such mixing can be achieved while still meeting the timing requirements of the time-critical traffic.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

b) gPTP specifies a media-independent sublayer that simplifies the integration within a single timing domain of multiple different networking technologies with radically different media access protocols. gPTP specifies a media-dependent sublayer for each medium. The information exchanged between PTP Instances has been generalized to support different packet formats and management schemes appropriate to the particular networking technology. IEEE Std 1588-2019, on the other

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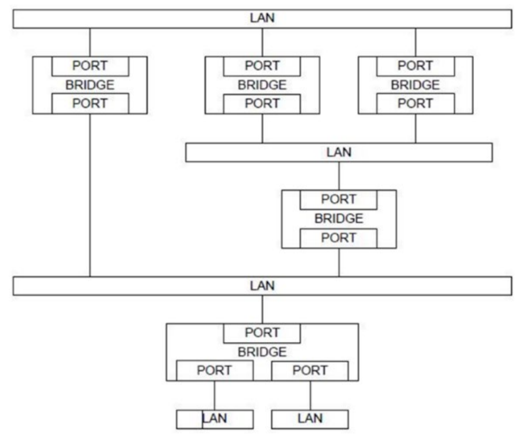

Figure 8-1—A Bridged Network

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

88. The methods practiced by Ribbon's use of the accused products include defining a common time reference for the device adapters, said common time reference including a frame of time having a plurality of time phases, each device adapter being uniquely assigned to one of said plurality of time phases, said plurality of time phases including a free-access phase. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The time aware bridged LAN uses 802.1AS base time to maintain synchronized time (common time reference) for all the ports in the bridges (device adapters). The bridges containing ports are scheduled for transmission of traffic based on the synchronized time. Each port associated with a specific set of transmission queues includes a plurality of transmission gates. A transmission gate can be in a closed state or open state. The functionality of assigned time phases is achieved using open gates transmitting data packets during transmission time. The functionality of free access phase is achieved when the gates are opened for transmission during any time. IEEE Std. 802.1AS™-2011 is normative and essential to implement an IEEE Std. 802.1Q Compliant System.

1.1 Project Number: P802.1Qbv
1.2 Type of Document: Standard
1.3 Life Cycle: Full Use

2.1 Title: Standard for Local and metropolitan area networks--Media Access Control (MAC) Bridges and Virtual Bridged Local Area Networks Amendment: Enhancements for Scheduled Traffic

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IEEE Std 802.1Q-2018
IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks

2. Normative references

The following referenced documents are indispensable for the application of this document (i.e., they must be understood and used, so each referenced document is cited in the text and its relationship to this document is explained). For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments or corrigenda) applies.

ANSI X3.159, American National Standards for Information Systems—Programming Language—C.[5]

IEEE Std 802®, IEEE Standard for Local and metropolitan area networks: Overview and Architecture.[6,7]

IEEE Std 802.1AB™, IEEE Standard for Local and metropolitan area networks—Station and Media Access Control Connectivity Discovery.

IEEE Std 802.1AC™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Service Definition.

IEEE Std 802.1AE™, IEEE Standard for Local and metropolitan area networks—Media Access Control (MAC) Security.

IEEE Std 802.1AS™, IEEE Standard for Local and metropolitan area networks—Timing and Synchronization for Time-Sensitive Applications in Bridged Local Area Networks.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

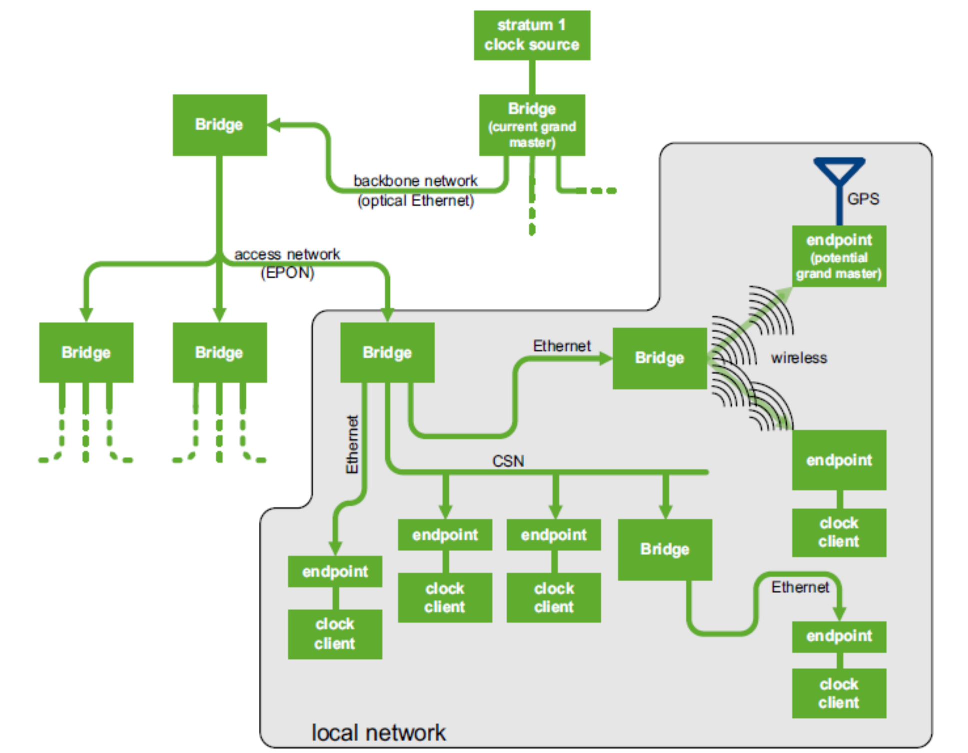

Figure 7-1—Time-aware network example

(Source: IEEE Standard for Local and metropolitan area networks - Timing and Synchronization for Time - Sensitive Applications in Bridged Local Area Networks - IEEE Std 802.1AS™-2011)

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b) For each queue corresponding to a numerically higher value of traffic class supported by the Port, the operation of the transmission selection algorithm supported by that queue determines that there is no frame available for transmission.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

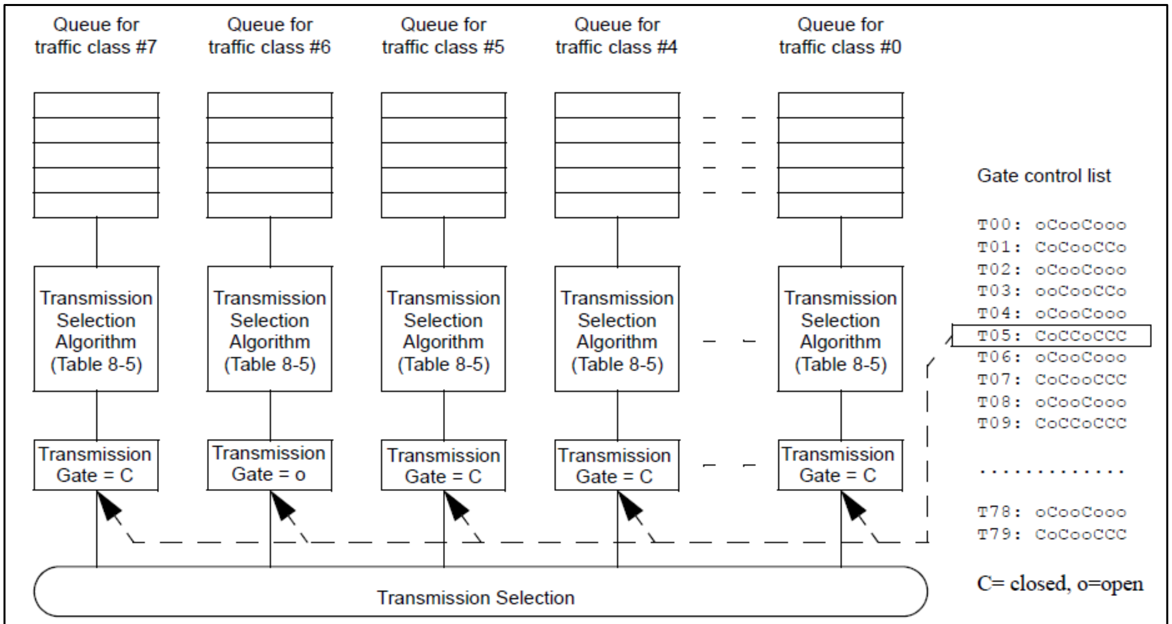


Figure 8-12—Transmission selection with gates

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

89. The methods practiced by Ribbon's use of the accused products include allowing a device adapter to transmit packets during said time phase uniquely assigned thereto and during said free-access phase. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. The open gates assigned to the transmission queue of a port (device adapter) transmit the data packets during the transmission time. The transmission selection algorithm determines the transmission time. If the gates are in permanent open state, the data packets are transmitted from the gates during any time. The functionality of assigned time phases

is achieved using open gates transmitting data packets during transmission time. The functionality of free access phase is achieved when the gates are permanently opened for transmission during any time.

8.6.8.4 Enhancements for scheduled traffic

A Bridge or an end station may support enhancements that allow transmission from each queue to be scheduled relative to a known timescale. In order to achieve this, a transmission gate is associated with each queue; the state of the transmission gate determines whether or not queued frames can be selected for transmission (see Figure 8-12). For a given queue, the transmission gate can be in one of two states:

a) *Open*: Queued frames are selected for transmission, in accordance with the definition of the transmission selection algorithm associated with the queue.
b) *Closed*: Queued frames are not selected for transmission.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

11.1.3 Transport of time-synchronization information

The transport of time-synchronization information by a PTP Instance, using Sync and Follow_Up (or just Sync) messages, is illustrated in Figure 11-2. The mechanism is mathematically equivalent to the mechanism described in IEEE Std 1588-2019 for a peer-to-peer transparent clock that is syntonized (see 11.4.5.1, 11.5.1, and 11.5.2.2 of IEEE Std 1588-2019). However, as will be seen shortly, the processes of transporting synchronization by a peer-to-peer transparent clock that is syntonized and by a boundary clock are mathematically and functionally equivalent. The main functional difference between the two types of clocks is that the boundary clock participates in best master selection and invokes the BMCA, while the peer-to-peer transparent clock does not participate in best master selection and does not invoke the BMCA (and implementations of the two types of clocks can be different).

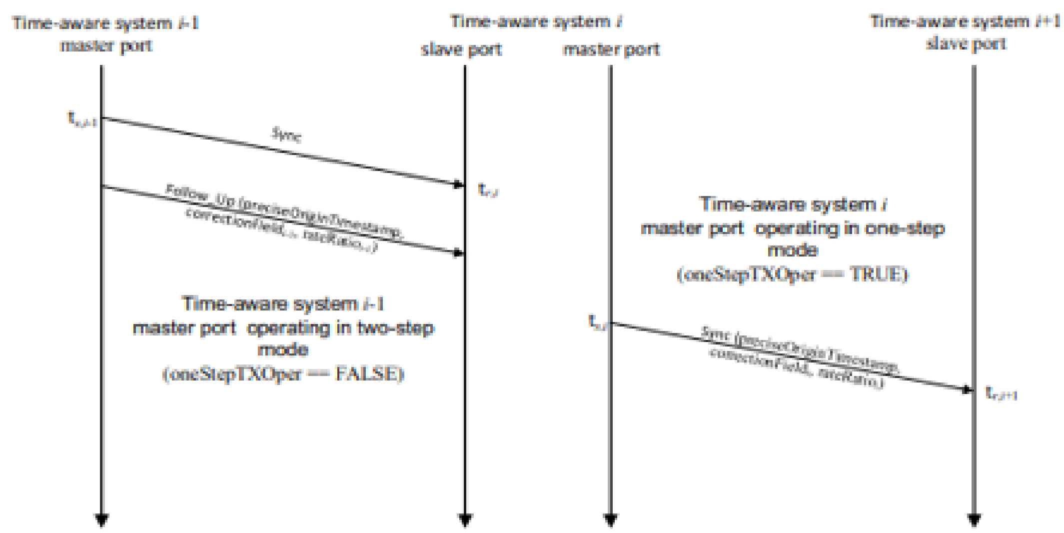


Figure 11-2—Transport of time-synchronization information

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

> A *gate control list* associated with each Port contains an ordered list of gate operations. Each gate operation changes the transmission gate state for the gate associated with each of the Port's traffic class queues. In an implementation that does not support enhancements for scheduled traffic, all gates are assumed to be permanently in the *open* state. Table 8-6 identifies the gate operation types, their parameters, and the actions that result from their execution. The state machines that control the execution of the gate control list, along with their variables and procedures, are specified in 8.6.9.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

90. The methods practiced by Ribbon's use of the accused products include cyclically repeating said frame. For example, the accused products are used by Ribbon to implement the IEEE 802.1Q standard. IEEE Std. 802.1Q-2018 supports cyclic queuing and forwarding structures to create synchronized frames and gates that are repeated periodically (Annex T).

Cyclic queuing and forwarding[53]

T.1 Overview of CQF

Cyclic queuing and forwarding (CQF) is a method of traffic shaping that can deliver deterministic, and easily calculated, latency for time-sensitive traffic streams. As the name implies, the principle underlying CQF is that stream traffic is transmitted and queued for transmission along a network path in a cyclic manner. Time is divided into numbered time intervals i, i+1, i+2, ... i+N, each of duration d. Frames transmitted by a Bridge, *Alice*, during time interval i are received by a downstream Bridge, *Bob*, during time interval i and are transmitted onwards by *Bob* towards Bridge *Charlie* during time interval i+1, and so on. A starting assumption is that, for a given traffic class, all Bridges and all end stations connected to a given bridge have a common understanding (to a known accuracy) of the start time of cycle i, and the cycle duration, d.

Frames transmitted by *Alice* during interval i are transmitted by *Bob* in interval i+1; the maximum possible delay experienced by a given frame is from the beginning of i to the end of i+1, or twice d. Similarly, the minimum possible delay experienced is from the end of i to the beginning of i+1, which is zero. More generally, the maximum delay experienced by a given frame is

$$(h+1) \times d$$

and the minimum delay experienced by a given frame is

$$(h-1) \times d$$

where h is the number of hops.

(Source: IEEE Standard for Local and Metropolitan Area Networks—Bridges and Bridged Networks- IEEE Std 802.1Q™-2018)

Traffic scheduling

This annex provides some background to the mechanisms provided in this standard for traffic scheduling, with the intent of providing some insight into the motivation for the provision of mechanisms for traffic scheduling and some of the ways that the mechanisms might be used.

Q.1 Motivation

Some applications have a need for frame delivery that is highly predictable in terms of the time at which frame transmission will occur, and the overall latency and jitter that will be experienced as the frame is propagated to its destination. Examples include industrial and automotive control applications, where data transmitted over the network is used to feed the parameters of control loops that are critical to the operation of the plant or machinery involved, and where frames carrying control data are transmitted on a repeating time schedule; late delivery of such frames can result in instability, inaccuracy, or failure of the operation of the control loops concerned. In some implementations, this need has been met by the provision of dedicated, highly engineered networks that are used solely for the transmission of time-critical control traffic; however, as the bandwidth occupied by such traffic is often low, and the cost of providing a dedicated control network can be high, it can be desirable to mix time-critical traffic with other classes of traffic in the same network, as long as such mixing can be achieved while still meeting the timing requirements of the time-critical traffic.

(Source: IEEE Standard for Local and metropolitan area networks— Bridges and Bridged Networks Amendment 25: Enhancements for Scheduled Traffic (IEEE 802.1Qbv-2015))

91. Ribbon has had knowledge of the '797 Patent at least as of the date when it was notified of the filing of this action.

92. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

93. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the '797 Patent.

COUNT VII

DIRECT INFRINGEMENT OF U.S. PATENT NO. 7,986,770

94. On July 26, 2011, United States Patent No. 7,986,770 ("the '770 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Method and Apparatus for Obtaining Telephone Status Over a Network."

95. Far North Patents is the owner of the '770 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '770 Patent against infringers, and to collect damages for all relevant times.

96. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Ribbon Kandy Smart Office and Ribbon Kandy Unified Communications as a Service families of products that include advanced presence information capabilities (collectively, "accused products").

(Source : https://www.ucaas-kandy.io/smart-office-clients)

Kandy Business Solutions leverage Ribbon's Compelling UC Client Architecture

Cloud UC buyers expect a sophisticated client experience. Kandy Cloud UC delivers Ribbon's next-generation unified communications clients, Smart Office. Visit our Unified Communication Clients pages for a detailed overview of why Smart Office clients are designed to be less expensive to deploy and manage than the competition. Our patent pending Omni technology enables Kandy Cloud UC customers to deliver and centrally manage their UCaaS experience without constantly managing client distribution and updates.

(Source : https://ribboncommunications.com/solutions/enterprise-solutions/uc-ucaas-cpaas/kandy-business-solutions-enterprises)

Smart Office clients deliver PBX calling services, unified company directory, user presence, instant messaging, conversation logs, voice and video conferencing. It's easy to find co-workers fast and get more done.

(Source : https://www.ucaas-kandy.io/smart-office-clients)

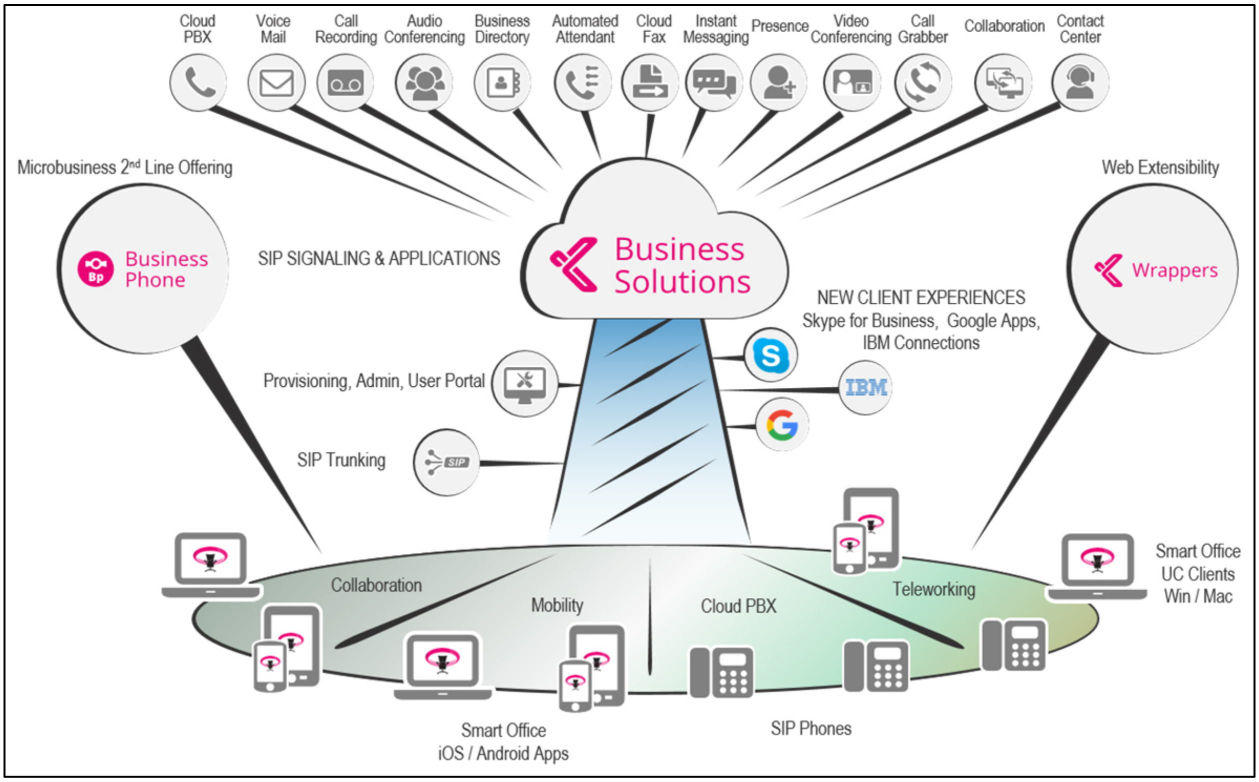
KANDY

Kandy For | Kandy Provides | Showcase | Ecosystem | Media | Company | Contact

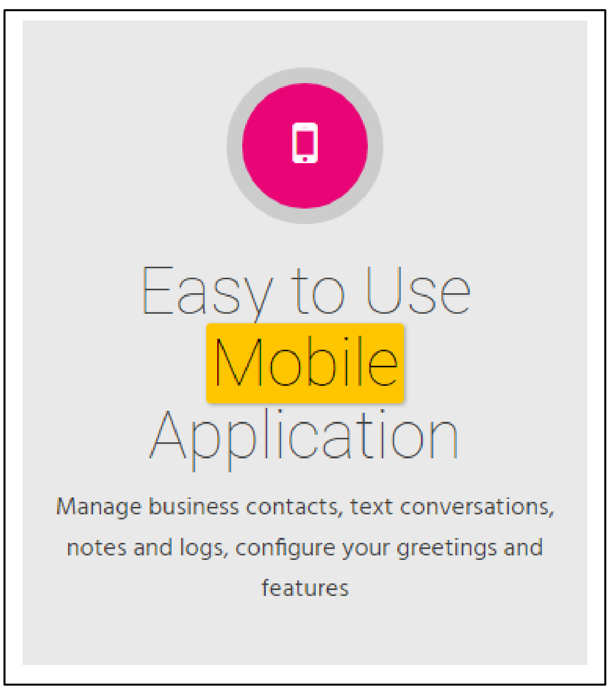
Home ▸ Kandy Provides ▸ Unified Communications as a Service - UCaaS ▸ UCaaS for Enterprises

It's the Right Time to Move Communications to the Cloud

(Source : https://www.kandy.io/kandy-provides/unified-communications-service-ucaas/ucaas-enterprises/)



(Source : https://www.kandy.io/kandy-provides/unified-communications-service-ucaas/ucaas-enterprises/)



(Source : https://www.kandy.io/kandy-provides/unified-communications-service-ucaas/ucaas-enterprises/)

Kandy delivers UC services from the cloud, including voice, video, IM, presence, collaboration and more. These services are coupled with our innovative Omni client technology that is uniquely capable of mashing up multiple cloud services and local applications into a single client experience. Omni technology consists of web programming with javascript, html, and style sheets, along with a device-specific client container that integrates with the local operating system. Omni provides significant cost benefits, user experience benefits and better alignment with cloud deployment of applications.

(Source : https://www.kandy.io/kandy-provides/unified-communications-service)

Of course, Kandy doesn't force contact center managers to abandon their existing investment. Kandy Wrappers can deliver UC services like chatbots and natural language-based AI agents with seamless escalation to live agents who can use voice, video or screen share to close the sale or close a case. Use Kandy APIs to add communication services to your website or mobile app.

(Source : https://www.kandy.io/kandy-provides/unified-communications-service)

What is Unified Communications: Functions

UC solutions minimally support real-time voice, video and text chat (instant messaging) capabilities. Most provide color-coded, presence-based directories (interactive user lists) that display real-time user status (i.e. on-line, off-line, or do-not-disturb). Most also offer PBX-like features such as hold, transfer, redial and three-way calling. Many offer advanced capabilities such as session recording, multiparty audio conferencing, multiparty video conferencing and screen sharing for hosting on-line meetings, presentations or product demonstrations. Often unified communication solutions also include a unified messaging component for voicemail and e-mail integration.

Many UC systems support APIs for integrating real-time communications capabilities into enterprise applications and business processes. Examples include the ability to call a customer directly from a CRM application or the ability to automatically direct an incoming call to the best employee based on skillset or presence state.

Unified Communications Components

A unified communications solution is composed of server-side and client-side components. The server-side component (sometimes referred to as an application server) is implemented as a traditional on-site IT solution or delivered as a cloud-based service. For the on-site case, some UC vendors offer turn-key UC server appliances, while others offer software-based solutions that run on industry-standard servers (typically x86 platforms running some version of Linux). Virtualized components are extremely common as they reduce the need for dedicated hardware and can simplify implementation; ultimately reducing the cost of the solution.

(Source : https://ribboncommunications.com/company/get-help/glossary/unified-communications-uc)

97. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 1 of the '770 Patent. Ribbon's infringement in this regard is ongoing.

98. Ribbon has infringed the '770 Patent by using the accused products and thereby practicing a method that includes monitoring, by a telephone status monitoring device, a status of a telephone via a first network, wherein the status includes an indication of whether the telephone is engaged in an active telephone call. For example, the accused products provide Smart Office Desktop and Mobile clients as a unified communications solution. Smart Office can be deployed with Kandy Business Solutions cloud. The Ribbon Application Server ("telephone status monitoring device") is a key enabler of Kandy Business Solutions. It integrates voice, video, instant messaging, presence, mobility, conferencing and collaboration over any network and any device. Smart Office clients delivers PBX calling services, a unified company directory, user presence, instant messaging, etc. Further, Smart Office allows user to move calls seamlessly across the devices using Call Grabber. Call Grabber allows the calls to be moved between desk phones, mobile clients or web clients. It is supported by the Smart Office Desktop & Mobile clients, Kandy Business Solutions and IP phones. The Kandy portal allows user to manage presence-based call management options. The presence of a user is still displayed to other users during an active call, even after moving the call to another device. The call status of all the Smart Office client devices is monitored by the Ribbon Application Server for tracking the presence information. For instance, when a user is engaged in a call using a desk phone that is located at an intra-site, the presence information of the desk phone is monitored via intranet ("first network").

DELIVER PHONE AND UC SERVICES TO COMPUTERS AND MOBILE DEVICES

Smart Office clients deliver rich collaboration experiences to PC, Mac, Apple, and Android devices; in or out of the office. Make and receive audio and video calls, send instant messages and collaborate with video and screen sharing directly from Windows or Mac devices. Seamlessly move calls between clients, desktop phones, and mobile devices. Employees can turn any workspace into a virtual office; ideal for telecommuters, mobile employees, and traditional office workers.

Smart Office clients deliver PBX calling services, unified company directory, user presence, instant messaging, conversation logs, voice and video conferencing. It's easy to find co-workers fast and get more done.

(Source: https://www.ucaas-kandy.io/smart-office-clients)

(Source: https://www.ucaas-kandy.io/smart-office-clients)

GENBAND Awarded Patent for Enterprise Unified Communications Technology - Call Grabber

Frisco, Texas,Feb. 11, 2014 – GENBAND, a leading developer of multimedia and cloud communications solutions, today announced the issuance of U.S. Patent number 8,600,006 for the one-step Unified Communications (UC) call transfer technology employed in GENBAND's Call Grabber enterprise technology. Deployed within the company's award-winning SMART OFFICE offerings, Call Grabber improves workforce mobility by enabling seamless, active call transfer between phones, smartphones, tablets and mobile devices in a one-step process that is undetectable to other parties on the call.

(Source: https://ribboncommunications.com/company/media-center/press-releases/genband-awarded-patent-enterprise-unified-communications-technology-call-grabber)

The Ribbon Application Server is the core of a Ribbon enterprise UC solution, it can be deployed stand-alone or in concert with a C20 Call Controller or CS 2100. It simplifies and enriches the user experience by providing fully-featured, IP-based unified communication applications. It integrates voice, video, instant messaging, presence, mobility, conferencing and collaboration over any network and any device, including mobile and tablet devices, WebRTC-based web clients, PCs, Macs, and IP phones. The Ribbon Application Server is SIP-based so it does not require proprietary business phones.

Most IP PBXs require call server hardware in multiple regional sites to support more than a few thousand users. The Ribbon Application Server's carrier heritage enables it to scale to two million users across geographically redundant servers. This massive scale is ideal for enterprises that are seeking a centralized private cloud deployment model to better leverage their data infrastructure investment. JITC certification assures the AS is secure and compelling for government deployments.

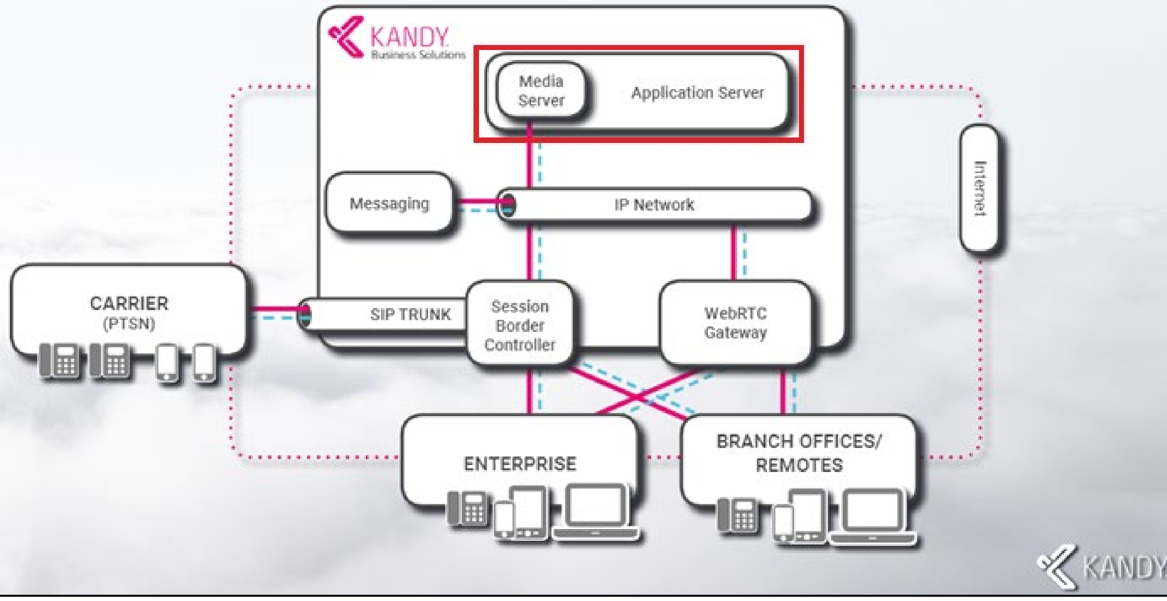
The Ribbon Telephony Application Server platform is one of the world's most widely-deployed voice and multimedia UC solutions, with over 27 million lines shipped. It is a key enabler of most of Ribbon's solutions including C20 Call Session Controller, Kandy Business Solutions, Smart Office hosted UC solutions as well as enterprise CS 2100 deployments.

(Source: https://ribboncommunications.com/solutions/enterprise-solutions/uc-ucaas-cpaas/premises-based-unified-communications)

Cloud-to-Cloud Connectivity

Kandy provides carrier-grade SBC and SIP Trunking services directly from the cloud, allowing organizations the ability to provide cloud-based telephony and unified communications services without the complexity of dealing with on-premises SBCs or media gateways. The (missing) is the simplest and most cost-effective option for most organizations because they realize the cost savings of a 100% cloud solution without additional hardware/software or maintenance expenses.

A 100% cloud solution doesn't translate to over utilizations of an organization's internet bandwidth. The Kandy SBC makes sure media for intra-site calls stay local. Only calls made to a recipient over the PSTN or made off-network go over the link to the Cloud



(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

Today we simply expect to be on the move, no one sits by the phone anymore, waiting for it to ring. Instead, we use multiple devices (desk phone, mobile phone, tablets PC/Macs and WebRTC-based browser clients) to stay connected. The problem is that too many of these devices can't actually talk to each other, forcing users to give out multiple phone numbers and forcing callers to guess which number is the right one to use at a speci ic moment.

Ribbon's Call Grabber feature, makes it easy to give out one number, answer it on the phone/device that is most convenient and seamlessly move the call to another device as the situation changes It's effortless to start a call at your desk and move it to your smartphone; there is no disruption or break in the conversation. Move calls multiple times between multiple devices, with just the touch of an icon or by entering a simple code. It even works with a home phone or a legacy (dumb) mobile phone.

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

Features

Call Grabber provides a variety of flexible options. Move calls to your mobile when you have to leave your desk or home without having to 'transfer' the call to your mobile number.

Calls can be moved between desk phones, smartphone clients, tablet clients, PC/Mac clients and web clients. Either a one-touch button or a short dial code initiates the action.

It supports mobile-initiated calls as well as fixed line-initiated calls.

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

Supported Platforms & Clients

- Ribbon Application Server
- Kandy Business Solutions
- Smart Office Desktop & Mobile Clients
- IP Phones
- Any legacy phone

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

ONE PHONE NUMBER SIMPLICITY

With Kandy & Smart Office, all of your devices and clients can be connected at the same time, staying connected to the office can be as simple as booting your computer. Use your business number to receive or make calls; you don't have to give out your personal mobile number to be accessible outside the office. And if you place calls from a Smart Office client your business number appears on the far end's caller ID. The Kandy cloud delivers a one number experience, across multiple devices. The Kandy Portal makes it easy to harness that power by choosing from a host of routing rules and time of day/ presence-based call management options. It's easy to choose how and when to stay connected and be productive; it's smart, it's Smart Office.

(Source: https://www.ucaas-kandy.io/smart-office-clients)

User Benefits

- Continue important calls when –
  - Leaving the home/office
  - Going to lunch
  - Running to printer, bathroom, break room or water cooler
  - Tracking down your boss or colleague
- Tracking down your boss or colleague
- Caller is unaware that call has been moved
- No delays or interruptions while call is moved
- Other users still see your phone "Presence" and know you are still on a call
- Supported on any type of mobile device
- Works irrespective of call origination
- Simple to use – with one touch menus or short dial codes.
- Instantly move the call to another device.
- Works with legacy phones too - just dial the Call Grabber number from the device to which you want the call to move and the call is moved

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

99. The methods practiced by Ribbon's use of the accused products include transmitting, by the telephone status monitoring device, the status of the telephone to another device via a second network, wherein the transmitting is in response to the another device attempting to place a call to the telephone, and wherein the first and the second networks are different networks. For example, Ribbon Application Server ("telephone status monitoring device") monitors and provides the presence information of a client device to other client devices. The Smart Office Call Grabber feature further supports both mobile initiated and fixed line-initiated calls. It allows the presence of the user to be displayed as in an active call, even after moving the call to another device. The presence information of the client devices at an intra-site (for instance, a desk phone) is monitored via intranet ("first network") and transmitted to other client devices (for instance, mobile phone) via internet ("second network"). For

example, when a user tries to place a call using a mobile phone, the user is provided with the presence information of the called party user using some presence indicators.

The Ribbon Application Server is the core of a Ribbon enterprise UC solution, it can be deployed stand-alone or in concert with a C20 Call Controller or CS 2100. It simplifies and enriches the user experience by providing fully-featured, IP-based unified communication applications. It integrates voice, video, instant messaging, presence, mobility, conferencing and collaboration over any network and any device, including mobile and tablet devices, WebRTC-based web clients, PCs, Macs, and IP phones. The Ribbon Application Server is SIP-based so it does not require proprietary business phones.

Most IP PBXs require call server hardware in multiple regional sites to support more than a few thousand users. The Ribbon Application Server's carrier heritage enables it to scale to two million users across geographically redundant servers. This massive scale is ideal for enterprises that are seeking a centralized private cloud deployment model to better leverage their data infrastructure investment. JITC certification assures the AS is secure and compelling for government deployments.

The Ribbon Telephony Application Server platform is one of the world's most widely-deployed voice and multimedia UC solutions, with over 27 million lines shipped. It is a key enabler of most of Ribbon's solutions including C20 Call Session Controller, Kandy Business Solutions, Smart Office hosted UC solutions as well as enterprise CS 2100 deployments.

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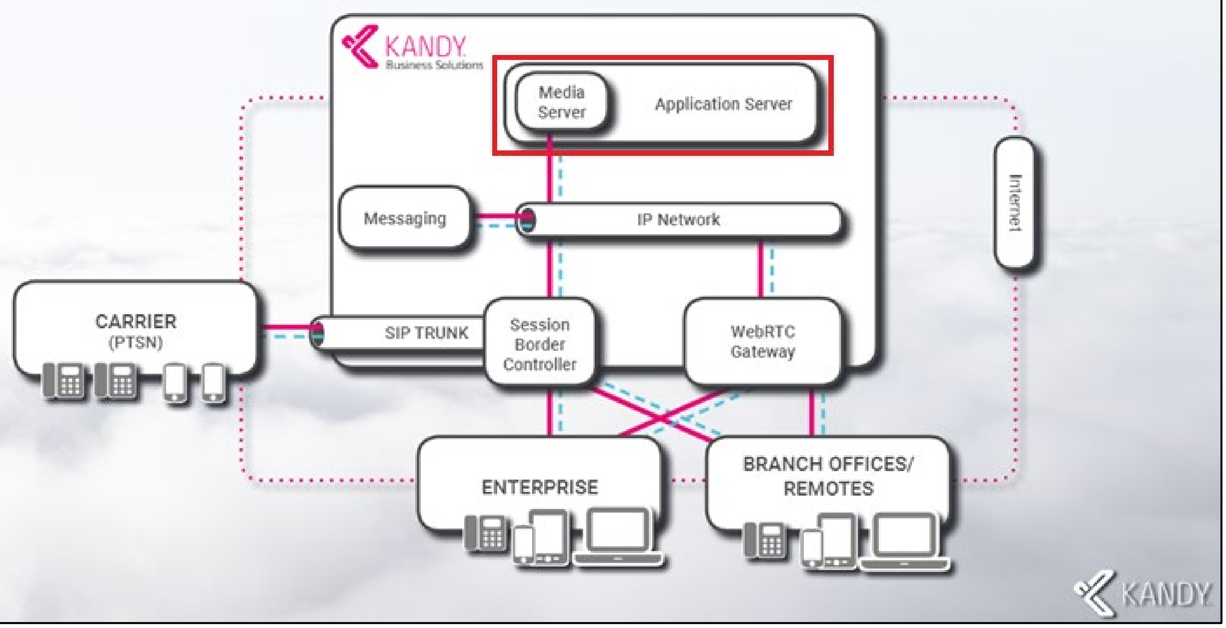

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

(Source: https://www.ucaas-kandy.io/smart-office-clients)

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With Kandy & Smart Office, all of your devices and clients can be connected at the same time, staying connected to the office can be as simple as booting your computer. Use your business number to receive or make calls; you don't have to give out your personal mobile number to be accessible outside the office. And if you place calls from a Smart Office client your business number appears on the far end's caller ID. The Kandy cloud delivers a one number experience, across multiple devices. The Kandy Portal makes it easy to harness that power by choosing from a host of routing rules and time of day/ presence-based call management options. It's easy to choose how and when to stay connected and be productive; it's smart, it's Smart Office.

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- Instantly move the call to another device.
- Works with legacy phones too - just dial the Call Grabber number from the device to which you want the call to move and the call is moved

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

100. Far North Patents only asserts method claims from the '770 Patent.

101. Ribbon has had knowledge of the '770 Patent at least as of the date when it was notified of the filing of this action.

102. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

103. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the '770 Patent.

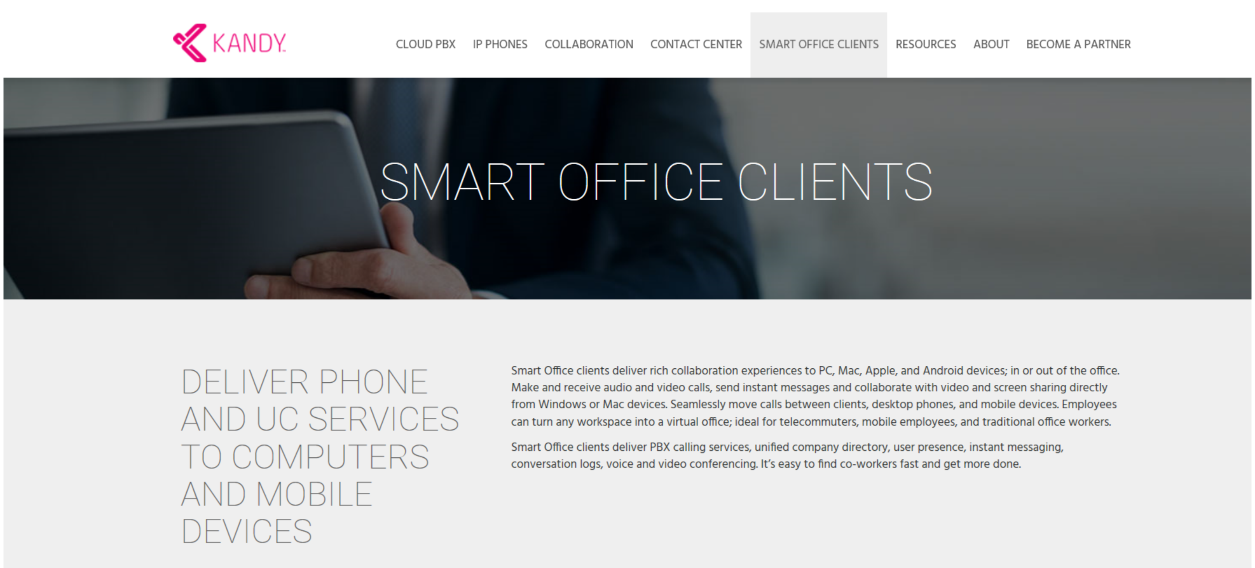
COUNT VIII

DIRECT INFRINGEMENT OF U.S. PATENT NO. 7,088,802

104. On August 8, 2006, United States Patent No. 7,088,802 ("the '802 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Method and Apparatus for Obtaining Telephone Status Over a Network."

105. Far North Patents is the owner of the '802 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '802 Patent against infringers, and to collect damages for all relevant times.

106. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Ribbon Kandy Smart Office and Ribbon Kandy Unified Communications as a Service families of products that include advanced presence information capabilities (collectively, "accused products"):



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Cloud UC buyers expect a sophisticated client experience. Kandy Cloud UC delivers Ribbon's next-generation unified communications clients, Smart Office. Visit our Unified Communication Clients pages for a detailed overview of why Smart Office clients are designed to be less expensive to deploy and manage than the competition. Our patent pending Omni technology enables Kandy Cloud UC customers to deliver and centrally manage their UCaaS experience without constantly managing client distribution and updates.

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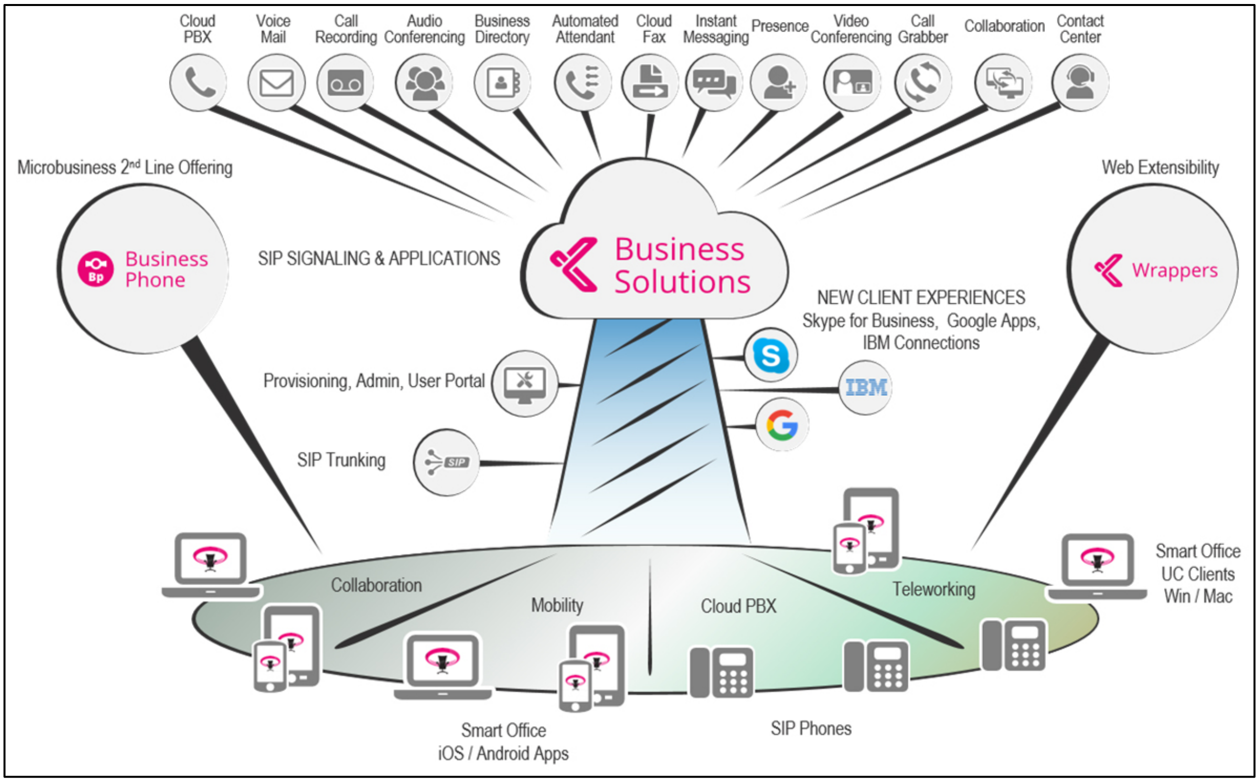
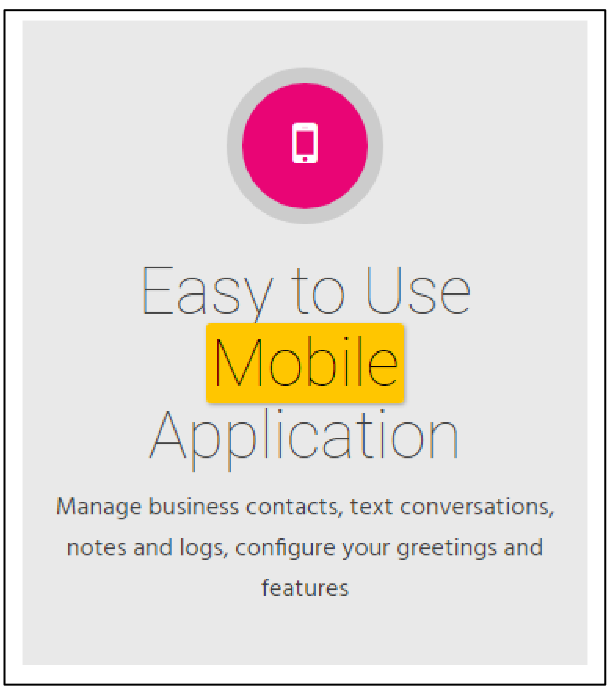
(Source : https://www.ucaas-kandy.io/smart-office-clients)

(Source : https://www.kandy.io/kandy-provides/unified-communications-service-ucaas/ucaas-enterprises/)



(Source : https://www.kandy.io/kandy-provides/unified-communications-service-ucaas/ucaas-enterprises/)



(Source : https://www.kandy.io/kandy-provides/unified-communications-service-ucaas/ucaas-enterprises/)

Kandy delivers UC services from the cloud, including voice, video, IM, presence, collaboration and more. These services are coupled with our innovative Omni client technology that is uniquely capable of mashing up multiple cloud services and local applications into a single client experience. Omni technology consists of web programming with javascript, html, and style sheets, along with a device-specific client container that integrates with the local operating system. Omni provides significant cost benefits, user experience benefits and better alignment with cloud deployment of applications.

(Source : https://www.kandy.io/kandy-provides/unified-communications-service)

Of course, Kandy doesn't force contact center managers to abandon their existing investment. Kandy Wrappers can deliver UC services like chatbots and natural language-based AI agents with seamless escalation to live agents who can use voice, video or screen share to close the sale or close a case. Use Kandy APIs to add communication services to your website or mobile app.

(Source : https://www.kandy.io/kandy-provides/unified-communications-service)

What is Unified Communications: Functions

UC solutions minimally support real-time voice, video and text chat (instant messaging) capabilities. Most provide color-coded, presence-based directories (interactive user lists) that display real-time user status (i.e. on-line, off-line, or do-not-disturb). Most also offer PBX-like features such as hold, transfer, redial and three-way calling. Many offer advanced capabilities such as session recording, multiparty audio conferencing, multiparty video conferencing and screen sharing for hosting on-line meetings, presentations or product demonstrations. Often unified communication solutions also include a unified messaging component for voicemail and e-mail integration.

Many UC systems support APIs for integrating real-time communications capabilities into enterprise applications and business processes. Examples include the ability to call a customer directly from a CRM application or the ability to automatically direct an incoming call to the best employee based on skillset or presence state.

Unified Communications Components

A unified communications solution is composed of server-side and client-side components. The server-side component (sometimes referred to as an application server) is implemented as a traditional on-site IT solution or delivered as a cloud-based service. For the on-site case, some UC vendors offer turn-key UC server appliances, while others offer software-based solutions that run on industry-standard servers (typically x86 platforms running some version of Linux). Virtualized components are extremely common as they reduce the need for dedicated hardware and can simplify implementation; ultimately reducing the cost of the solution.

(Source : https://ribboncommunications.com/company/get-help/glossary/unified-communications-uc)

107. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 5 of the '802 Patent. Ribbon's infringement in this regard is ongoing.

108. Ribbon has infringed the '802 Patent by using the accused products and thereby practicing a process for determining wireless telecommunication device status including accessing a wireless telecommunication device status file associated with a called party over a network and monitoring the status of the called party's wireless telecommunication device and providing that device status to the device status file. For example, the accused products provide Smart Office Desktop and Mobile clients as a unified communications solution. Smart Office can be deployed with Kandy Business Solutions cloud. Ribbon Application Server is a key enabler of Kandy Business Solutions. It integrates voice, video, instant messaging, presence, mobility, conferencing and collaboration over any network and any device. Smart Office clients delivers PBX calling services, unified company directory, user presence, instant messaging, etc. Further, Smart Office allows user to move calls seamlessly across the devices using Call Grabber. Call Grabber allows the calls to be moved between desk phones, mobile clients (wireless telecommunications devices) or web clients. It is supported by the Smart Office Desktop & Mobile clients, Kandy Business Solutions and IP phones. The Kandy portal allows a user to manage presence-based call management options. The presence of a user is still displayed to other users during an active call, even after moving the call to another device. The call status of all the Smart Office client devices is monitored (accessing) by the Ribbon Application Server for tracking the presence information. The presence information is stored in a device status file for providing it to other client devices in the environment. For instance, when a user is engaged in a call using a mobile client, the presence information of the mobile client is monitored and stored at a device status file.

DELIVER PHONE AND UC SERVICES TO COMPUTERS AND MOBILE DEVICES

Smart Office clients deliver rich collaboration experiences to PC, Mac, Apple, and Android devices; in or out of the office. Make and receive audio and video calls, send instant messages and collaborate with video and screen sharing directly from Windows or Mac devices. Seamlessly move calls between clients, desktop phones, and mobile devices. Employees can turn any workspace into a virtual office; ideal for telecommuters, mobile employees, and traditional office workers.

Smart Office clients deliver PBX calling services, unified company directory, user presence, instant messaging, conversation logs, voice and video conferencing. It's easy to find co-workers fast and get more done.

(Source: https://www.ucaas-kandy.io/smart-office-clients)

(Source: https://www.ucaas-kandy.io/smart-office-clients)

GENBAND Awarded Patent for Enterprise Unified Communications Technology - Call Grabber

Frisco, Texas,Feb. 11, 2014 – GENBAND, a leading developer of multimedia and cloud communications solutions, today announced the issuance of U.S. Patent number 8,600,006 for the one-step Unified Communications (UC) call transfer technology employed in GENBAND's Call Grabber enterprise technology. Deployed within the company's award-winning SMART OFFICE offerings, Call Grabber improves workforce mobility by enabling seamless, active call transfer between phones, smartphones, tablets and mobile devices in a one-step process that is undetectable to other parties on the call.

(Source: https://ribboncommunications.com/company/media-center/press-releases/genband-awarded-patent-enterprise-unified-communications-technology-call-grabber)

The Ribbon Application Server is the core of a Ribbon enterprise UC solution, it can be deployed stand-alone or in concert with a C20 Call Controller or CS 2100. It simplifies and enriches the user experience by providing fully-featured, IP-based unified communication applications. It integrates voice, video, instant messaging, presence, mobility, conferencing and collaboration over any network and any device, including mobile and tablet devices, WebRTC-based web clients, PCs, Macs, and IP phones. The Ribbon Application Server is SIP-based so it does not require proprietary business phones.

Most IP PBXs require call server hardware in multiple regional sites to support more than a few thousand users. The Ribbon Application Server's carrier heritage enables it to scale to two million users across geographically redundant servers. This massive scale is ideal for enterprises that are seeking a centralized private cloud deployment model to better leverage their data infrastructure investment. JITC certification assures the AS is secure and compelling for government deployments.

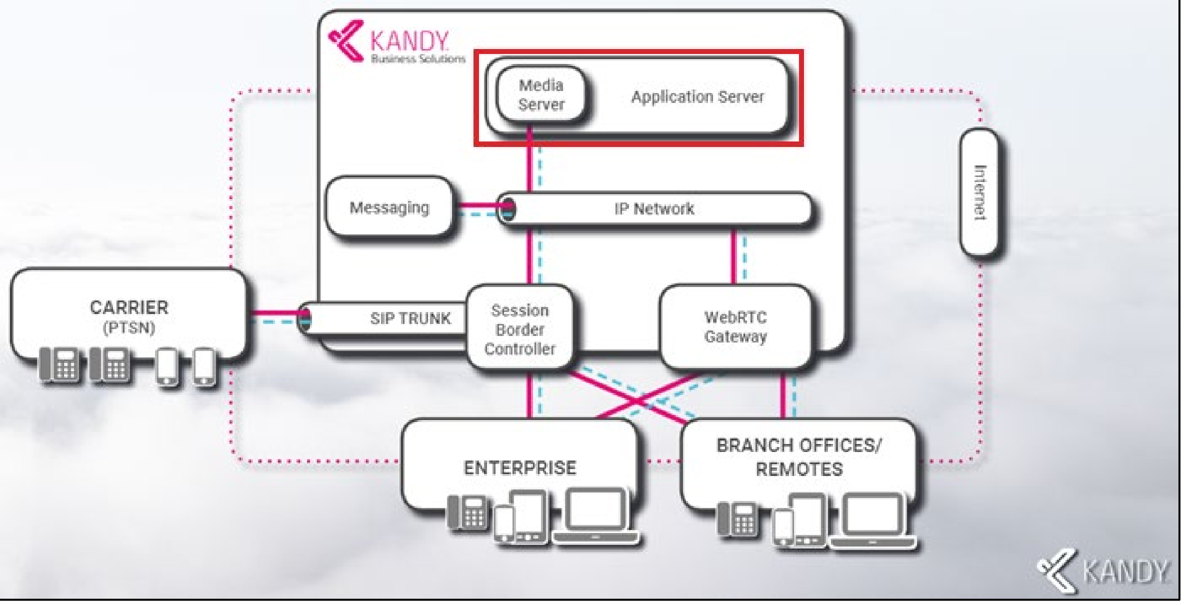
The Ribbon Telephony Application Server platform is one of the world's most widely-deployed voice and multimedia UC solutions, with over 27 million lines shipped. It is a key enabler of most of Ribbon's solutions including C20 Call Session Controller, Kandy Business Solutions, Smart Office hosted UC solutions as well as enterprise CS 2100 deployments.

(Source: https://ribboncommunications.com/solutions/enterprise-solutions/uc-ucaas-cpaas/premises-based-unified-communications)

Cloud-to-Cloud Connectivity

Kandy provides carrier-grade SBC and SIP Trunking services directly from the cloud, allowing organizations the ability to provide cloud-based telephony and unified communications services without the complexity of dealing with on-premises SBCs or media gateways. The (missing) is the simplest and most cost-effective option for most organizations because they realize the cost savings of a 100% cloud solution without additional hardware/software or maintenance expenses.

A 100% cloud solution doesn't translate to over utilizations of an organization's internet bandwidth. The Kandy SBC makes sure media for intra-site calls stay local. Only calls made to a recipient over the PSTN or made off-network go over the link to the Cloud



(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

Today we simply expect to be on the move, no one sits by the phone anymore, waiting for it to ring. Instead, we use multiple devices (desk phone, mobile phone, tablets PC/Macs and WebRTC-based browser clients) to stay connected. The problem is that too many of these devices can't actually talk to each other, forcing users to give out multiple phone numbers and forcing callers to guess which number is the right one to use at a speci ic moment.

Ribbon's Call Grabber feature, makes it easy to give out one number, answer it on the phone/device that is most convenient and seamlessly move the call to another device as the situation changes It's effortless to start a call at your desk and move it to your smartphone; there is no disruption or break in the conversation. Move calls multiple times between multiple devices, with just the touch of an icon or by entering a simple code. It even works with a home phone or a legacy (dumb) mobile phone.

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

Features

Call Grabber provides a variety of flexible options. Move calls to your mobile when you have to leave your desk or home without having to 'transfer' the call to your mobile number.

Calls can be moved between desk phones, smartphone clients, tablet clients, PC/Mac clients and web clients. Either a one-touch button or a short dial code initiates the action.

It supports mobile-initiated calls as well as fixed line-initiated calls.

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

Supported Platforms & Clients

- Ribbon Application Server
- Kandy Business Solutions
- Smart Office Desktop & Mobile Clients
- IP Phones
- Any legacy phone

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

ONE PHONE NUMBER SIMPLICITY

With Kandy & Smart Office, all of your devices and clients can be connected at the same time, staying connected to the office can be as simple as booting your computer. Use your business number to receive or make calls; you don't have to give out your personal mobile number to be accessible outside the office. And if you place calls from a Smart Office client your business number appears on the far end's caller ID. The Kandy cloud delivers a one number experience, across multiple devices. The Kandy Portal makes it easy to harness that power by choosing from a host of routing rules and time of day/ presence-based call management options. It's easy to choose how and when to stay connected and be productive; it's smart, it's Smart Office.

(Source: https://www.ucaas-kandy.io/smart-office-clients)

User Benefits

- Continue important calls when –
  - Leaving the home/office
  - Going to lunch
  - Running to printer, bathroom, break room or water cooler
  - Tracking down your boss or colleague
- Tracking down your boss or colleague
- Caller is unaware that call has been moved
- No delays or interruptions while call is moved
- Other users still see your phone "Presence" and know you are still on a call
- Supported on any type of mobile device
- Works irrespective of call origination
- Simple to use – with one touch menus or short dial codes.
- Instantly move the call to another device.
- Works with legacy phones too - just dial the Call Grabber number from the device to which you want the call to move and the call is moved

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

109. The processes practiced by Ribbon's use of the accused products include sending a calling party a status message comprising the status of the called party's wireless telecommunication device, wherein the status message is sent via a pager. For example, Ribbon Application Server monitors and provides the presence information of a client device to other client devices. The Smart Office Call Grabber feature further supports both mobile initiated and fixed line initiated calls. It allows the presence of the user to be displayed as in an active call, even after moving the call to another device. The presence information of a client devices (for instance, a mobile client) would be monitored and transmitted to other client devices (for instance, desktop client) and displayed as some presence indicators on the application's UI. For example, when a user tries to place a call using a desktop client, the user will be provided with

the presence information of the called party user using some presence indicators. The status is also presented on a display on a cellular phone, indicating that the status is sent via a pager.

The Ribbon Application Server is the core of a Ribbon enterprise UC solution, it can be deployed stand-alone or in concert with a C20 Call Controller or CS 2100. It simplifies and enriches the user experience by providing fully-featured, IP-based unified communication applications. It integrates voice, video, instant messaging, presence, mobility, conferencing and collaboration over any network and any device, including mobile and tablet devices, WebRTC-based web clients, PCs, Macs, and IP phones. The Ribbon Application Server is SIP-based so it does not require proprietary business phones.

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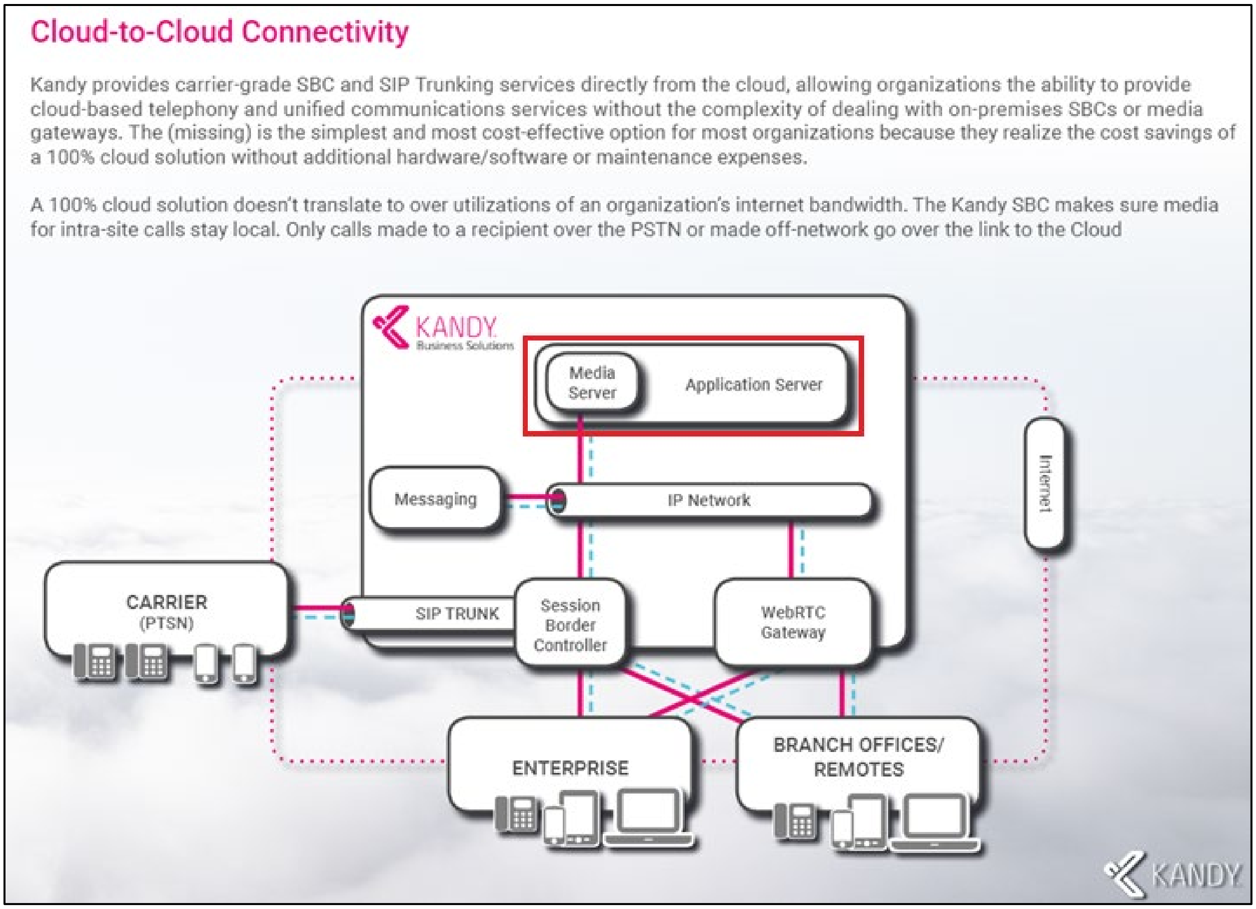
(Source: https://ribboncommunications.com/solutions/enterprise-solutions/uc-ucaas-cpaas/premises-based-unified-communications)



(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

(Source: https://www.ucaas-kandy.io/smart-office-clients)

Features

Call Grabber provides a variety of flexible options. Move calls to your mobile when you have to leave your desk or home without having to 'transfer' the call to your mobile number.

Calls can be moved between desk phones, smartphone clients, tablet clients, PC/Mac clients and web clients. Either a one-touch button or a short dial code initiates the action.

It supports mobile-initiated calls as well as fixed line-initiated calls.

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(Source: https://www.ucaas-kandy.io/smart-office-clients)

User Benefits

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- Works with legacy phones too - just dial the Call Grabber number from the device to which you want the call to move and the call is moved

(Source: https://www.juxto.com/info/Call-Grabber-Data-sheet.pdf)

110. Far North Patents only asserts method claims from the '802 Patent.

111. Ribbon has had knowledge of the '802 Patent at least as of the date when it was notified of the filing of this action.

112. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

113. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the '802 Patent.

COUNT IX

DIRECT INFRINGEMENT OF U.S. PATENT NO. 7,149,517

114. On December 12, 2006, United States Patent No. 7,149,517 ("the '517 Patent") was duly and legally issued by the United States Patent and Trademark Office for an invention entitled "Methods and Systems for Implementation of the Calling Name Delivery Service Through Use of a Location Register in a Network Element in a Wireless Network."

115. Far North Patents is the owner of the '517 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the '517 Patent against infringers, and to collect damages for all relevant times.

116. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Sonus SBC 7000 SBC Core and Sonus Voice Exchange (VX) Series families of products, that include calling name services for mobile phones (collectively, "accused products"):


Sonus CCG TELECOM

PRODUCTS ABOUT CONTACT US

SBC 7000

HOME / PRODUCT OVERVIEW / SBC 7000

SBC 7000

EASILY SCALE ACROSS GLOBAL MARKETS

Service providers and large enterprises are facing a torrent of insatiable demand for multimedia communications applications across global markets. The foundational strength of the Sonus SBC 7000 is its ability to support high capacity requirements for real-time, multimedia Session Initiation Protocol (SIP) traffic.

The SBC 7000 can scale up to 150,000 sessions via simple software licenses, allowing operational teams to expand capacity in days, not months.

(Source : http://sonussuperstore.com/featured-products/sbc-7000/)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source : https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Introducing the VX Product Suite

Introduction

Sonus' **Voice Exchange (VX) Series** is the industry's first fully-integrated multi-service voice switch with Any-to-Any gateway functionality, creating a next-generation solution for enterprise VoIP enablement.

- Enterprise VoIP and telephony integration for Unified Messaging and Unified Communications
- Integrated, multi-function voice switch
- Intelligent IP to IP Mediation
- Vendor Independent Remote Survivability
- Extensive call system interoperability
- True telecom traffic management
- Advanced call security
- Expandable platform with optional software modules
- Enables intelligent, flexible, lower cost VoIP and Mobility

(Source : https://support.sonus.net/display/VXDOC52/Introducing+the+VX+Product+Suite)

Features Added in VX Release 4.3

- Basic RTCP Voice over IP Metrics
- NCG Relay
- SIP 180 Ringing Event Pass Through and Longer SIP Call-ID
  - Events Pass Through
  - Longer SIP Call-Id
- PBX2 Certification Enhancements
  - VLAN Tagging
  - PBX Network Scenario
- MWI Enhancements
- QSIG Calling Party Name Support
- Passing Call Diversion Information Between QSIG and SIP

(Source : https://support.sonus.net/download/attachments/6488120/VXDOC-4.7.2-July2010.pdf?api=v2)

QSIG Calling Party Name Support

This feature adds the Calling Party Name Identification Presentation supplementary service (SS-CNIP) in QSIG.

This feature is supported both for calls from QSIG network to the SIP network (incoming call) as well as for a call from the SIP network to QSIG network (outgoing call).

This feature does not support the user profiles which enable the user to override the Calling Name Identification Restriction (Refer [1], section6.2.2.2).

The basic call services are already supported in QSIG. This feature will add a supplementary service which enables the called party to see the name of the calling party.

The Calling Party Name Identification support is already implemented in the NI-2 flavor of ISDN protocol supported on the VX. This feature shall implement the same for QSIG.

Calling Name Identification Presentation (CNIP) is a supplementary service which is offered to the called user and which provides the name of the calling user (Calling Party Name) to the called user.

(Source : https://support.sonus.net/download/attachments/6488120/VXDOC-4.7.2-July2010.pdf?api=v2)

117. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 7 of the ‘517 Patent. Ribbon’s infringement in this regard is ongoing.

118. Ribbon has infringed the ‘517 Patent by using the accused products and thereby practicing a method for a location register (LR) of a network element in a wireless network to provide information associated with a wireless unit including causing the LR to include a plurality of entries with each entry including an identifier and corresponding respectively to a wireless unit, each identifier of an entry being associated with information corresponding

respectively to the wireless unit. For example, the accused products provide an ENUM CNAM feature. ENUM CNAM provides calling name information associated with a wireless unit. A calling number (identifier) is used for forming an ENUM query. Using this ENUM query, calling name information is obtained from ENUM database (location register). Further, SBC core 7000 supports multiple wireless devices and subscribers.

What Is an SBC?

A Session Border Controller is a special-purpose device that protects and regulates IP communications flows. As the name implies, session border controllers are deployed at network borders to control IP communications sessions. Originally conceived to protect and control VoIP networks, SBCs are now used to regulate all forms of real-time communications including VoIP, IP video, text chat and collaboration sessions..

(Source: https://support.sonus.net/display/SBXDOC81/About+the+SBC+Core+Family)

The Ribbon SBC Portfolio is comprised of the following Session Border Controller products:

SBC Edge

- SBC 1000/2000

SBC Core

- SBC 5000 series
- SBC 5400
- SBC 7000
- SBC Software Edition (SWe)
- SBC SWe Cloud

(Source: https://support.sonus.net/display/SBXDOC81/About+the+SBC+Core+Family)

- **SBC 7000 Series**
  Targets large session count deployments (up to 150,000 sessions). These capacities make this product particularly well suited for large service providers. Example deployment scenarios include:
  - Service Provider Access – High subscriber and simultaneous call scale coupled with high availability and redundancy.
  - Service Provider Peering – High simultaneous call scale coupled with high availability and redundancy.
  - Enterprise and Service Provider Video – Supports large WAN interface bandwidth.
  - Wireless – Supports a large number of subscribers and calls where high availability is essential.

(Source: https://support.sonus.net/display/SBXDOC71/About+SBC+Core)

Telephone Number Mapping (ENUM) transforms the telephone number into a universal identifier recognized across different devices and applications (voice, fax, mobile, e-mail, text messaging, location-based services and the Internet). ENUM can also be used to implement a variety of services like number portability, call forwarding etc. It uses a DNS NAPTR record to translate a telephone number into a Uniform Resource Identifier or IP address that can be used in Internet communications.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

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The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Userinfo Parameters in ENUM Response

SBC Core supports transparently passing userinfo parameters received in the ENUM Naming Authority Pointer Response (NAPTR) response in the outgoing INVITE Request Uniform Resource Identifier (R-URI). The SBC transparently sends custom and standard userinfo parameters such as tgrp, trunk-context, cic and rn received in ENUM response. The SBC sends these parameters in the R-URI of the egress INVITE.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

119. The methods practiced by Ribbon's use of the accused products include causing the LR in response to receipt of a query including the identifier to use the identifier to find an entry having the identifier in common with the query. For example, a calling number is used to form an ENUM search query. The query is used to find user information from the database and an ENUM response is received with required information.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

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SBC Core supports transparently passing userinfo parameters received in the ENUM Naming Authority Pointer Response (NAPTR) response in the outgoing INVITE Request Uniform Resource Identifier (R-URI). The SBC transparently sends custom and standard userinfo parameters such as tgrp, trunk-context, cic and rn received in ENUM response. The SBC sends these parameters in the R-URI of the egress INVITE.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

120. The methods practiced by Ribbon's use of the accused products include, based on the information being associated with the identifier of the entry, causing the LR to retrieve the information corresponding to the wireless unit and causing the LR to provide the information in a response to the query. For example, a query including a calling number is used to find user information from an ENUM database, and an ENUM response is received with the required information.

Telephone Number Mapping (ENUM) transforms the telephone number into a universal identifier recognized across different devices and applications (voice, fax, mobile, e-mail, text messaging, location-based services and the Internet). ENUM can also be used to implement a variety of services like number portability, call forwarding etc. It uses a DNS NAPTR record to translate a telephone number into a Uniform Resource Identifier or IP address that can be used in Internet communications.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database When the ENUM returns NAPTR resource records, the SBC adds the caller name.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Userinfo Parameters in ENUM Response

SBC Core supports transparently passing userinfo parameters received in the ENUM Naming Authority Pointer Response (NAPTR) response in the outgoing INVITE Request Uniform Resource Identifier (R-URI). The SBC transparently sends custom and standard userinfo parameters such as tgrp, trunk-context, cic and rn received in ENUM response. The SBC sends these parameters in the R-URI of the egress INVITE.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

121. Far North Patents only asserts method claims from the ‘517 Patent.

122. Ribbon has had knowledge of the ‘517 Patent at least as of the date when it was notified of the filing of this action.

123. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

124. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the ‘517 Patent.

COUNT X

DIRECT INFRINGEMENT OF U.S. PATENT NO. 6,618,588

125. On September 9, 2003, United States Patent No. 6,618,588 (“the ‘588 Patent”) was duly and legally issued by the United States Patent and Trademark Office for an invention entitled “Methods and Systems for Implementation of the Calling Name Delivery Service Through Use of a Location Register in a Network Element in a Wireless Network.”

126. Far North Patents is the owner of the ‘588 Patent, with all substantive rights in and to that patent, including the sole and exclusive right to prosecute this action and enforce the ‘588 Patent against infringers, and to collect damages for all relevant times.

127. Ribbon made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Sonus SBC 7000 SBC Core and Sonus Voice Exchange (VX) Series families of products, that include calling name services for mobile phones (collectively, “accused products”):

Sonus CCG TELECOM

PRODUCTS ABOUT CONTACT US

SBC 7000

HOME / PRODUCT OVERVIEW / SBC 7000

SBC 7000

EASILY SCALE ACROSS GLOBAL MARKETS

Service providers and large enterprises are facing a torrent of insatiable demand for multimedia communications applications across global markets. The foundational strength of the Sonus SBC 7000 is its ability to support high capacity requirements for real-time, multimedia Session Initiation Protocol (SIP) traffic.

The SBC 7000 can scale up to 150,000 sessions via simple software licenses, allowing operational teams to expand capacity in days, not months.

(Source : http://sonussuperstore.com/featured-products/sbc-7000/)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source : https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Introducing the VX Product Suite

Introduction

Sonus' **Voice Exchange (VX) Series** is the industry's first fully-integrated multi-service voice switch with Any-to-Any gateway functionality, creating a next-generation solution for enterprise VoIP enablement.

- Enterprise VoIP and telephony integration for Unified Messaging and Unified Communications
- Integrated, multi-function voice switch
- Intelligent IP to IP Mediation
- Vendor Independent Remote Survivability
- Extensive call system interoperability
- True telecom traffic management
- Advanced call security
- Expandable platform with optional software modules
- Enables intelligent, flexible, lower cost VoIP and Mobility

(Source : https://support.sonus.net/display/VXDOC52/Introducing+the+VX+Product+Suite)

Features Added in VX Release 4.3

- Basic RTCP Voice over IP Metrics
- NCG Relay
- SIP 180 Ringing Event Pass Through and Longer SIP Call-ID
  - Events Pass Through
  - Longer SIP Call-Id
- PBX2 Certification Enhancements
  - VLAN Tagging
  - PBX Network Scenario
- MWI Enhancements
- QSIG Calling Party Name Support
- Passing Call Diversion Information Between QSIG and SIP

(Source : https://support.sonus.net/download/attachments/6488120/VXDOC-4.7.2-July2010.pdf?api=v2)

QSIG Calling Party Name Support

This feature adds the Calling Party Name Identification Presentation supplementary service (SS-CNIP) in QSIG.

This feature is supported both for calls from QSIG network to the SIP network (incoming call) as well as for a call from the SIP network to QSIG network (outgoing call).

This feature does not support the user profiles which enable the user to override the Calling Name Identification Restriction (Refer [1], section6.2.2.2).

The basic call services are already supported in QSIG. This feature will add a supplementary service which enables the called party to see the name of the calling party.

The Calling Party Name Identification support is already implemented in the NI-2 flavor of ISDN protocol supported on the VX. This feature shall implement the same for QSIG.

Calling Name Identification Presentation (CNIP) is a supplementary service which is offered to the called user and which provides the name of the calling user (Calling Party Name) to the called user.

(Source : https://support.sonus.net/download/attachments/6488120/VXDOC-4.7.2-July2010.pdf?api=v2)

128. By doing so, Ribbon has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 5 of the ‘588 Patent. Ribbon’s infringement in this regard is ongoing.

129. Ribbon has infringed the ‘588 Patent by using the accused products and thereby practicing a method for a location register (LR) of a network element in a wireless network to provide information associated with a wireless unit including causing the LR to include a plurality of entries with each entry including a mobile identity number (MIN) and/or a mobile directory number (MDN) and corresponding respectively to a wireless unit, each MIN and/or

MDN of an entry being associated with information corresponding respectively to the wireless unit, wherein the information comprises a presentation allowance with respect to a name, the MIN and/or the MDN. For example, the accused products provide an ENUM CNAM feature. ENUM CNAM provides calling name information associated with a wireless unit. A calling number (MIN) is used for forming an ENUM query. Using this ENUM query, calling name information is obtained from ENUM database (location register). Further, SBC core 7000 supports multiple wireless devices and subscribers. SBC core 7000 also supports 'caller privacy feature' (presentation allowance), which helps in restricting caller number's information. For example, as can be seen from below citation, when 'Caller Privacy' feature is enabled corresponding to a user/MIN it can have Caller ID presentation restriction and, in that case, the username would be sent as 'anonymous user'.

What Is an SBC?

A Session Border Controller is a special-purpose device that protects and regulates IP communications flows. As the name implies, session border controllers are deployed at network borders to control IP communications sessions. Originally conceived to protect and control VoIP networks, SBCs are now used to regulate all forms of real-time communications including VoIP, IP video, text chat and collaboration sessions..

(Source: https://support.sonus.net/display/SBXDOC81/About+the+SBC+Core+Family)

The Ribbon SBC Portfolio is comprised of the following Session Border Controller products:

SBC Edge

- SBC 1000/2000

SBC Core

- SBC 5000 series
- SBC 5400
- SBC 7000
- SBC Software Edition (SWe)
- SBC SWe Cloud

(Source: https://support.sonus.net/display/SBXDOC81/About+the+SBC+Core+Family)

- **SBC 7000 Series**
  Targets large session count deployments (up to 150,000 sessions). These capacities make this product particularly well suited for large service providers. Example deployment scenarios include:
  - Service Provider Access – High subscriber and simultaneous call scale coupled with high availability and redundancy.
  - Service Provider Peering – High simultaneous call scale coupled with high availability and redundancy.
  - Enterprise and Service Provider Video – Supports large WAN interface bandwidth.
  - Wireless – Supports a large number of subscribers and calls where high availability is essential.

(Source: https://support.sonus.net/display/SBXDOC71/About+SBC+Core)

Telephone Number Mapping (ENUM) transforms the telephone number into a universal identifier recognized across different devices and applications (voice, fax, mobile, e-mail, text messaging, location-based services and the Internet). ENUM can also be used to implement a variety of services like number portability, call forwarding etc. It uses a DNS NAPTR record to translate a telephone number into a Uniform Resource Identifier or IP address that can be used in Internet communications.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Userinfo Parameters in ENUM Response

SBC Core supports transparently passing userinfo parameters received in the ENUM Naming Authority Pointer Response (NAPTR) response in the outgoing INVITE Request Uniform Resource Identifier (R-URI). The SBC transparently sends custom and standard userinfo parameters such as tgrp, trunk-context, cic and rn received in ENUM response. The SBC sends these parameters in the R-URI of the egress INVITE.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

The SBC Core supports caller privacy dynamically based on privacy header.

The special URI *anonymous@host; user=phone* in From header field denotes an anonymous user whose presentation is restricted. Microsoft Lync 2010 Mediation Server uses the alias name of the user in the FROM header (instead of phone number) and without *user=phone* parameter, to hide a phone number. The SBC identifies these two cases as presentation restriction. When the presentation is restricted, if the SIP trunk has a default number configured, the SBC uses that number in the user part of FROM header while sending the egress message.

The SBC Core supports caller privacy for incoming and outgoing calls as described below:

- For incoming calls from the SIP Trunk that the Enterprise SBC is to present to the Mediation Server, if the Calling Party Number/ANI indicates restricted presentation, the From URI in the SIP INVITE sent to the Mediation Server denotes an anonymous user.
- For outgoing calls from the Mediation Server, if the From URI in the SIP INVITE sent to the Enterprise SBC denotes an anonymous user or is a SIP URI with an alias name, the SBC excludes or sets the default value to Calling Party Number sent to the SIP Trunk.

(Source: https://support.sonus.net/display/SBXDOC81/Caller+Privacy+Support)

130. The methods practiced by Ribbon's use of the accused products include causing the LR in response to receipt of a query including the MIN and/or the MDN to use the MIN and/or the MDN to find an entry having the MIN and/or the MDN in common with the query and checking that the information associated with the MIN and/or the MDN of the entry comprises the presentation allowance. For example, a calling number is used to form an ENUM search query. In case of wireless devices, a mobile number (MIN) is a calling number. The query is used to find user information from the database and an ENUM response is received with required information.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Userinfo Parameters in ENUM Response

SBC Core supports transparently passing userinfo parameters received in the ENUM Naming Authority Pointer Response (NAPTR) response in the outgoing INVITE Request Uniform Resource Identifier (R-URI). The SBC transparently sends custom and standard userinfo parameters such as tgrp, trunk-context, cic and rn received in ENUM response. The SBC sends these parameters in the R-URI of the egress INVITE.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

The SBC Core supports caller privacy dynamically based on privacy header.

The special URI *anonymous@host; user=phone* in From header field denotes an anonymous user whose presentation is restricted. Microsoft Lync 2010 Mediation Server uses the alias name of the user in the FROM header (instead of phone number) and without *user=phone* parameter, to hide a phone number. The SBC identifies these two cases as presentation restriction. When the presentation is restricted, if the SIP trunk has a default number configured, the SBC uses that number in the user part of FROM header while sending the egress message.

The SBC Core supports caller privacy for incoming and outgoing calls as described below:

- For incoming calls from the SIP Trunk that the Enterprise SBC is to present to the Mediation Server, if the Calling Party Number/ANI indicates restricted presentation, the From URI in the SIP INVITE sent to the Mediation Server denotes an anonymous user.
- For outgoing calls from the Mediation Server, if the From URI in the SIP INVITE sent to the Enterprise SBC denotes an anonymous user or is a SIP URI with an alias name, the SBC excludes or sets the default value to Calling Party Number sent to the SIP Trunk.

(Source: https://support.sonus.net/display/SBXDOC81/Caller+Privacy+Support)

131. The methods practiced by Ribbon's use of the accused products include, based on the information being associated with the MIN and/or the MDN of the entry, causing the LR to retrieve the information corresponding to the wireless unit and causing the LR to provide the information in a response to the query. For example, a query including a calling number is used to find user information from the database and an ENUM response is received from the database with required information. The desired ENUM response is obtained based on the information associated with user such as 'Caller Privacy' feature. When this feature is enabled corresponding to a user/MIN it can have Caller ID presentation restriction and, in that case, the username is sent as 'anonymous user'.

Telephone Number Mapping (ENUM) transforms the telephone number into a universal identifier recognized across different devices and applications (voice, fax, mobile, e-mail, text messaging, location-based services and the Internet). ENUM can also be used to implement a variety of services like number portability, call forwarding etc. It uses a DNS NAPTR record to translate a telephone number into a Uniform Resource Identifier or IP address that can be used in Internet communications.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

ENUM CNAM

ENUM CNAM enables calling name lookups on the IP network without having to consult an external element in the SS7 network, thus simplifying call routing and network operations.

To provision ENUM CNAM, specify the trigger criteria in the Number Translation Criteria entity and specify the ENUM CNAM as the lookup type. ENUM CNAM, provisioned through the DNS-ENUM Service Definition screen, uses the same triggering and escape criteria as SCP-based CNAM. When a Routing Request triggers an ENUM CNAM lookup, the SBC uses the calling number to form a domain name and puts it in an ENUM CNAM query to the ENUM database. When the ENUM returns NAPTR resource records, the SBC adds the caller name.

The ERE platform supports number translations based on the host part (contains a fully-qualified domain name) of the SIP headers. As the IP address format is not supported for host part lookup, it is recommended to use the fully-qualified domain name. The SBC retrieves the destination domain from the various SIP headers and sends it to the ERE as part of the policy request. The domain name lookup is case insensitive and partial matches are supported, while first priority is always for the full match.

The supported destination domain SIP headers are:

- INVITE Message, `R-URI Header`
- INVITE Message, `TO Header`
- 3XX Message, `Contact Header`

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

Userinfo Parameters in ENUM Response

SBC Core supports transparently passing userinfo parameters received in the ENUM Naming Authority Pointer Response (NAPTR) response in the outgoing INVITE Request Uniform Resource Identifier (R-URI). The SBC transparently sends custom and standard userinfo parameters such as tgrp, trunk-context, cic and rn received in ENUM response. The SBC sends these parameters in the R-URI of the egress INVITE.

(Source: https://support.sonus.net/display/SBXDOC81/ENUM+Support)

The SBC Core supports caller privacy dynamically based on privacy header.

The special URI *anonymous@host; user=phone* in From header field denotes an anonymous user whose presentation is restricted. Microsoft Lync 2010 Mediation Server uses the alias name of the user in the FROM header (instead of phone number) and without *user=phone* parameter, to hide a phone number. The SBC identifies these two cases as presentation restriction. When the presentation is restricted, if the SIP trunk has a default number configured, the SBC uses that number in the user part of FROM header while sending the egress message.

The SBC Core supports caller privacy for incoming and outgoing calls as described below:

- For incoming calls from the SIP Trunk that the Enterprise SBC is to present to the Mediation Server, if the Calling Party Number/ANI indicates restricted presentation, the From URI in the SIP INVITE sent to the Mediation Server denotes an anonymous user.
- For outgoing calls from the Mediation Server, if the From URI in the SIP INVITE sent to the Enterprise SBC denotes an anonymous user or is a SIP URI with an alias name, the SBC excludes or sets the default value to Calling Party Number sent to the SIP Trunk.

(Source: https://support.sonus.net/display/SBXDOC81/Caller+Privacy+Support)

132. Far North Patents only asserts method claims from the ‘588 Patent.

133. Ribbon has had knowledge of the ‘588 Patent at least as of the date when it was notified of the filing of this action.

134. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

135. Far North Patents and/or its predecessors-in-interest have satisfied all statutory obligations required to collect pre-filing damages for the full period allowed by law for infringement of the ‘588 Patent.

ADDITIONAL ALLEGATIONS REGARDING INFRINGEMENT

136. Ribbon has also indirectly infringed the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent by inducing others to directly infringe the ‘105 Patent, the ‘437

Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent. Ribbon has induced the end-users, Ribbon’s customers, to directly infringe (literally and/or under the doctrine of equivalents) the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent by using the accused products.

137. Ribbon took active steps, directly and/or through contractual relationships with others, with the specific intent to cause them to use the accused products in a manner that infringes one or more claims of the patents-in-suit, including, for example, Claims 1 and 23 of the ‘105 Patent, Claim 9 of the ‘437 Patent, Claim 1 of the ‘230 Patent, Claims 1 and 14 of the ‘053 Patent, Claim 27 of the ‘702 Patent, Claim 30 of the ‘797 Patent, Claim 1 of the ‘770 Patent, Claim 5 of the ‘802 Patent, Claim 7 of the ‘517 Patent, and Claim 5 of the ‘588 Patent.

138. Such steps by Ribbon included, among other things, advising or directing customers and end-users to use the accused products in an infringing manner; advertising and promoting the use of the accused products in an infringing manner; and/or distributing instructions that guide users to use the accused products in an infringing manner.

139. Ribbon has performed these steps, which constitute induced infringement, with the knowledge of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent and with the knowledge that the induced acts constitute infringement.

140. Ribbon was and is aware that the normal and customary use of the accused products by Ribbon’s customers would infringe the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent. Ribbon’s inducement is ongoing.

141. Ribbon has also induced its affiliates, or third-party manufacturers, shippers, distributors, retailers, or other persons acting on its or its affiliates' behalf, to directly infringe (literally and/or under the doctrine of equivalents) the '105 Patent, the '437 Patent, the '230 Patent, the '053 Patent, the '702 Patent, the '797 Patent, the '770 Patent, the '802 Patent, the '517 Patent, and the '588 Patent by importing, selling or offering to sell the accused products.

142. Ribbon has at least a significant role in placing the accused products in the stream of commerce in Texas and elsewhere in the United States.

143. Ribbon directs or controls the making of accused products and their shipment to the United States, using established distribution channels, for sale in Texas and elsewhere within the United States.

144. Ribbon directs or controls the sale of the accused products into established United States distribution channels, including sales to nationwide retailers.

145. Ribbon's established United States distribution channels include one or more United States based affiliates and divisions.

146. Ribbon directs or controls the sale of the accused products in nationwide retailers, including for sale in Texas and elsewhere in the United States, and expects and intends that the accused products will be so sold.

147. Ribbon took active steps, directly and/or through contractual relationships with others, with the specific intent to cause such persons to import, sell, or offer to sell the accused products in a manner that infringes one or more claims of the patents-in-suit, including, for example, Claims 1 and 23 of the '105 Patent, Claim 9 of the '437 Patent, Claim 1 of the '230 Patent, Claims 1 and 14 of the '053 Patent, Claim 27 of the '702 Patent, Claim 30 of the '797

Patent, Claim 1 of the ‘770 Patent, Claim 5 of the ‘802 Patent, Claim 7 of the ‘517 Patent, and Claim 5 of the ‘588 Patent.

148. Such steps by Ribbon included, among other things, making or selling the accused products outside of the United States for importation into or sale in the United States, or knowing that such importation or sale would occur; and directing, facilitating, or influencing its affiliates, or third-party manufacturers, shippers, distributors, retailers, or other persons acting on its or their behalf, to import, sell, or offer to sell the accused products in an infringing manner.

149. Ribbon performed these steps, which constitute induced infringement, with the knowledge of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent and with the knowledge that the induced acts would constitute infringement.

150. Ribbon performed such steps in order to profit from the eventual sale of the accused products in the United States.

151. Ribbon’s inducement is ongoing.

152. Ribbon has also indirectly infringed by contributing to the infringement of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent. Ribbon has contributed to the direct infringement of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent by the end-user of the accused products.

153. The accused products have special features that are specially designed to be used in an infringing way and that have no substantial uses other than ones that infringe the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the

‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent, including, for example, Claims 1 and 23 of the ‘105 Patent, Claim 9 of the ‘437 Patent, Claim 1 of the ‘230 Patent, Claims 1 and 14 of the ‘053 Patent, Claim 27 of the ‘702 Patent, Claim 30 of the ‘797 Patent, Claim 1 of the ‘770 Patent, Claim 5 of the ‘802 Patent, Claim 7 of the ‘517 Patent, and Claim 5 of the ‘588 Patent.

154. The special features include advanced quality monitoring capabilities, advanced quality of service capabilities, advanced presence information capabilities, and calling name services for mobile phones, used in a manner that infringes the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent.

155. The special features constitute a material part of the invention of one or more of the claims of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent and are not staple articles of commerce suitable for substantial non-infringing use.

156. Ribbon’s contributory infringement is ongoing.

157. Furthermore, Ribbon has a policy or practice of not reviewing the patents of others (including instructing its employees to not review the patents of others), and thus has been willfully blind of Far North Patents’ patent rights. *See, e.g.*, M. Lemley, “Ignoring Patents,” 2008 Mich. St. L. Rev. 19 (2008).

158. Ribbon’s actions are at least objectively reckless as to the risk of infringing valid patents and this objective risk was either known or should have been known by Ribbon.

159. Ribbon has knowledge of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent.

160. Ribbon’s customers have infringed the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent.

161. Ribbon encouraged its customers’ infringement.

162. Ribbon’s direct and indirect infringement of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent is, has been, and/or continues to be willful, intentional, deliberate, and/or in conscious disregard of Far North Patents’ rights under the patents.

163. Far North Patents has been damaged as a result of the infringing conduct by Ribbon alleged above. Thus, Ribbon is liable to Far North Patents in an amount that adequately compensates it for such infringements, which, by law, cannot be less than a reasonable royalty, together with interest and costs as fixed by this Court under 35 U.S.C. § 284.

JURY DEMAND

Far North Patents hereby requests a trial by jury on all issues so triable by right.

PRAYER FOR RELIEF

Far North Patents requests that the Court find in its favor and against Ribbon, and that the Court grant Far North Patents the following relief:

a. Judgment that one or more claims of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, the ‘053 Patent, the ‘702 Patent, the ‘797 Patent, the ‘770 Patent, the ‘802 Patent, the ‘517 Patent, and the ‘588 Patent have been infringed, either literally and/or under the doctrine of equivalents, by Ribbon and/or all others acting in concert therewith;

b. A permanent injunction enjoining Ribbon and its officers, directors, agents, servants, affiliates, employees, divisions, branches, subsidiaries, parents, and all others acting in concert therewith from infringement of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, and the ‘053 Patent; or, in the alternative, an award of a reasonable ongoing royalty for future infringement of the ‘105 Patent, the ‘437 Patent, the ‘230 Patent, and the ‘053 Patent by such entities;

c. Judgment that Ribbon account for and pay to Far North Patents all damages to and costs incurred by Far North Patents because of Ribbon’s infringing activities and other conduct complained of herein, including an award of all increased damages to which Far North Patents is entitled under 35 U.S.C. § 284;

d. That Far North Patents be granted pre-judgment and post-judgment interest on the damages caused by Ribbon’s infringing activities and other conduct complained of herein;

e. That this Court declare this an exceptional case and award Far North Patents its reasonable attorney’s fees and costs in accordance with 35 U.S.C. § 285; and

f. That Far North Patents be granted such other and further relief as the Court may deem just and proper under the circumstances.

Dated: December 26, 2019

Respectfully submitted,

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