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

FAR NORTH PATENTS, LLC, Plaintiff, v. MITEL NETWORKS CORPORATION, and MITEL NETWORKS, INC., Defendants.	CIVIL ACTION NO. 4:19-cv-942 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 Mitel Networks Corporation and Mitel Networks, Inc., (collectively "Mitel" 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 Mitel Networks Corporation is a corporation organized and existing under the laws of Canada, with a place of business at 350 Legget Dr., Kanata, ON K2K 2W7.

3. Defendant Mitel Networks, Inc. is a corporation organized and existing under the laws of Delaware. Mitel maintains a regional office in the Eastern District of Texas located at either or both of 5360 Legacy Drive, Suite 300, Plano, TX 75024-3130 or 5850 Granite Pkwy,

Suite 600, Plano, TX 75024. Mitel Networks, Inc. may be served through its registered agent, CT Corporation System, at 1999 Bryan Street, Suite 900, Dallas, TX 75201.

4. The Defendants identified in paragraphs 2-3 above (collectively, "Mitel") are companies which together comprise one of the world's largest entities specializing in communications technologies.

5. The Mitel 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 Mitel 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 Mitel 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 Mitel pursuant to due process and/or the Texas Long Arm Statute because, *inter alia*, (i) Mitel has done and continues to do business in Texas; (ii) Mitel 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, and (iii) Mitel Networks, Inc. is registered to do business in Texas. In addition, or in the alternative, this Court has personal jurisdiction over Mitel Networks Corporation pursuant to Fed. R. Civ. P. 4(k)(2).

10. Venue is proper as to Defendant Mitel Networks Corporation, which is organized under the laws of Canada. 28 U.S.C. § 1391(c)(3) provides that "a defendant not resident in the United States may be sued in any judicial district, and the joinder of such a defendant shall be disregarded in determining where the action may be brought with respect to other defendants."

11. Venue is proper in this district as to Mitel Networks, Inc. pursuant to 28 U.S.C. § 1400(b). Venue is further proper because Mitel 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. Mitel also has regular and established places of business in this district, including at either or both of 5360 Legacy Drive, Suite 300, Plano, TX 75024-3130 or 5850 Granite Pkwy, Suite 600, Plano, TX 75024 (as shown in the below screenshots from Mitel's website, https://www.mitel.com/contact/locations/texas-plano and from a search of the Collin CAD site).

TEXAS-PLANO

Address
5360 Legacy Drive Suite 300
Plano, TX 75024-3130
United States
Get Directions >

Phone: +1 469 574 8000

Hours
Monday 8AM-5PM
Tuesday 8AM-5PM
Wednesday 8AM-5PM
Thursday 8AM-5PM
Friday 8AM-5PM
Saturday Closed
Sunday Closed

7	**2681973** P-9000-213-1167-1	MITEL NETWORKS INC	5850 Granite Pkwy #00600 Plano, TX 75024	BPP at 5850 Granite Pkwy	$3,364,830

BACKGROUND

12. 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”).

13. 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.

14. 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).

15. 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.

16. 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.

17. 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.

18. 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.

19. 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.

20. 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.

21. 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

22. 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."

23. 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.

24. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel Performance Analytics, Mitel Enterprise Manager, and Mitel Connect families of products that include advanced quality monitoring capabilities (collectively, "accused products").

Mitel Performance Analytics

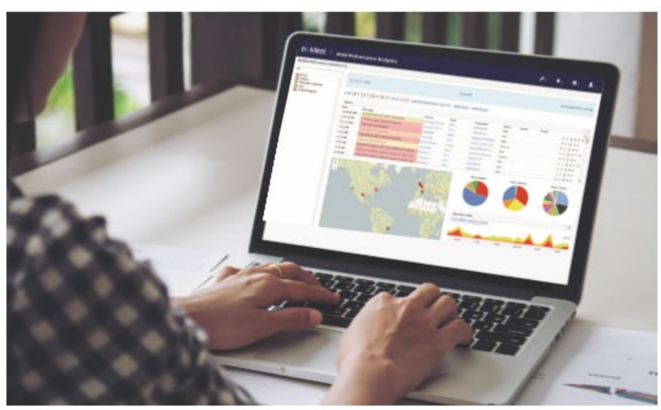
Available with Mitel Premium Software Assurance

Key Benefits

- Faster problem detection and resolution
- Simplified management of large networks
- Improved user satisfaction and adoption
- Better use of IT resources

Why MPA?

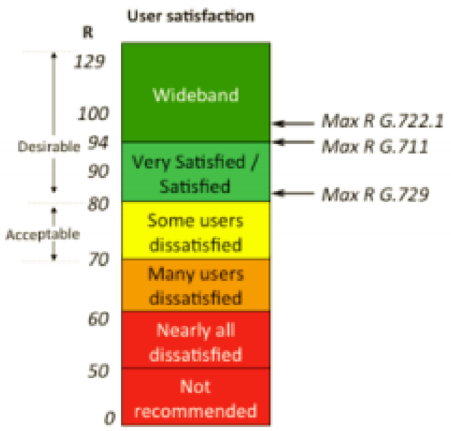
- Intuitive, multi-tenant data rich dashboards
- Comprehensive testing tools
- Network diagram integration
- Reporting tools that add value



(Source : https://www.mitel.com/-/media/mitel/pdf/brochures/brochure_mpa_3_0.pdf?modified=20191016161501)

R VALUE

Mitel Performance Analytics uses the Extended E-model to estimate the received voice quality for VoIP calls. The E-model is based on ITU-T Recommendation G.107. The primary output of the E-model calculations is a scalar quality rating value known as the R value. The R scale ranges from 0 (bad) to a maximum of 129 (excellent). The range for R is 6.5 to 93.4 for narrowband codecs (G.729 and G.711) to a theoretical maximum of 129 for wideband codecs (G722.1).



(Source: https://martellotech.com/wp-content/uploads/2018/07/MPA_2.1_System_Guide.pdf)

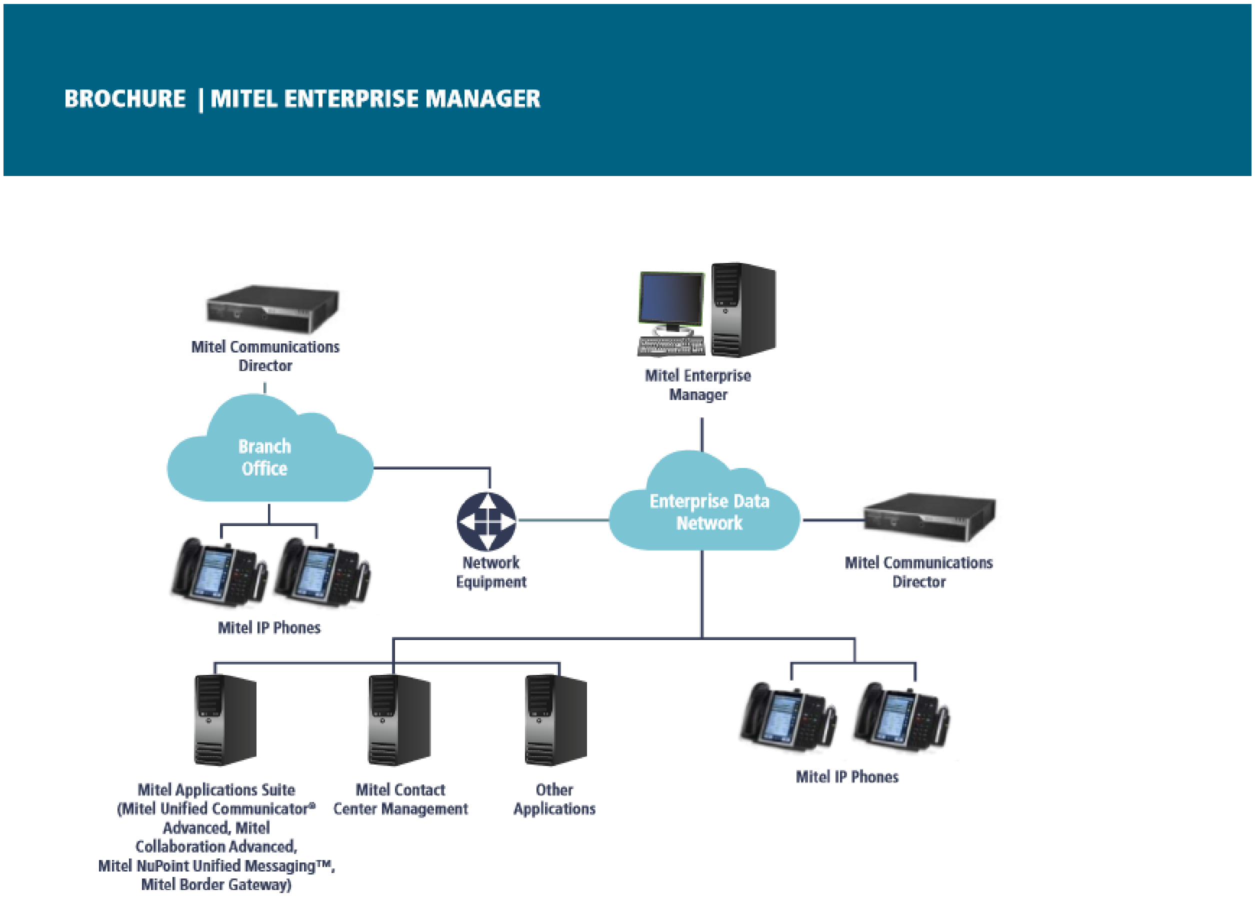
COLOR CODING FOR VOICE QUALITY AND SIP VOICE QUALITY

The voice quality information is color coded to enable rapid identification of trends. Mitel Performance Analytics uses the following thresholds.

		R VALUE	
COLOR	VOICE QUALITY	FROM	TO
Green	Good	80	93.2
Yellow	Adequate	70	80
Orange	Poor	60	70
Red	Bad	0	60

For good quality VoIP, the R value should be 80 or better. An R value less than 70 indicates poor audio quality and less than 60 is generally unusable.

(Source: https://martellotech.com/wp-content/uploads/2018/07/MPA_2.1_System_Guide.pdf)



(Source :

https://www.cacommunications.com/productcatalog/mitel/pdf/MitelEnterpriseManager-Brochure.pdf)

Performance management is about optimizing system operation under normal conditions to proactively monitor and even correct potential problems before they occur. With regard to IP communications, any problems with the performance or configuration of the IP networking equipment tends to result in poor voice quality. In other words, the IP phones get the blame when in most cases poor voice quality is not the fault of the phone, but of the underlying IP network. However, the burden of proof tends to fall on the application and associated devices in which the performance problem manifests itself. This is why voice quality is the first performance management application to be supported within Mitel® Enterprise Manager.

Mitel Voice Quality Management Related Capabilities

Enterprise Manager, as well as our platform systems and associated IP phones, provides for multiple levels of voice quality monitoring and troubleshooting capabilities. Basic voice quality management tools are built into the platform systems and IP phones for monitoring and troubleshooting of solutions at an individual system level. For ongoing monitoring of voice quality over a network of multiple systems, Enterprise Manager offers the Voice Quality Manager (VQM) option. Enterprise Manager VQM provides a network-wide view of voice quality including drill-

Mitel utilizes a hierarchical approach to monitor all calls in a multi-system network. Each of the individual Mitel 3300 IP Communications Platform (ICP) systems collects statistics from their attached phones. The VQM option within Enterprise Manager collects the call quality statistics from all the 3300 ICP systems and provides a single point for integration with external management systems. For troubleshooting voice quality or other IP network problems, Enterprise Manager includes diagnostic tools for monitoring network interfaces, performing remote trace routes and

(Source :

http://www.ashtelecom.co.uk/imgpdf/92_378_Voice_Quality_Performance_Management.pdf)

Rating Voice Quality

All Mitel IP phones[2] collect and report voice quality statistics suitable for use in generating voice quality scores such as R-Value or Mean Opinion Score (MOS). Both R-Value and MOS are methods for rating user satisfaction levels for telephony voice quality. The R-Value is a quality rating factor that is the primary computational output from the ITU-T G.107 recommendation for transmission planning known as the E-Model. R-Value can be used as an input into a Mean Opinion Score (MOS) computation, that attempts to assess conversational voice quality taking into account human opinion factors associated with various types and levels of impairments.

The Enterprise Manager reporting option[3] produces voice quality reports using the R-Value scale to provide a zero to 100 quality rating with 95 being the highest rating possible for a G.711 codec-based VoIP call. Many more specialized VoIP performance management tools such as Viola NetAlly apply additional analysis to take into account human opinion factors to arrive at a MOS score. A MOS score is a subjective value from 1 (lowest) to 5 (highest) that rates the voice quality of a call and should reflect an average of a large sampling of user opinions obtained on calls with known impairments. There are many potential algorithms and variations that can be used for computing a MOS score for an IP telephony call. There is no standard MOS algorithm to use at this time and MOS scoring cannot be directly compared unless the same algorithm is used. The table below provides a general comparison between the R-value and MOS scales, though it is just an estimation.

R-Value	MOS	User Satisfaction
90-100	4.34-4.5	Very Satisfied
80-89	4.03-4.33	Satisfied
70-79	3.60-4.02	Some users dissatisfied
60-69	3.10-3.59	Many users dissatisfied
<59	<3.10	Nearly all users dissatisfied

(Source :

http://www.ashtelecom.co.uk/imgpdf/92_378_Voice_Quality_Performance_Management.pdf)

Powering connections

Mitel Connect System Administration Guide

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

Monitoring Call Quality

The Call Quality page enables easy troubleshooting of problems related to call and network quality by providing access to records of every media stream that occurs in the Mitel Connect system. The call quality metrics are derived from monitoring IP network impairments such as packet loss and delay.

Aspects of Call Quality

Call Quality is evaluated based on thresholds for packet loss, delay/latency, and jitter.

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

Table 191: Fields on the Call Quality Details Tab (Continued)

Field Name	Description
Codec	The audio codec and sample rate used in the stream If more than one codec is used for a call, only the last codec used for the call is displayed.
MOS	The stream's MOS, which the Monitoring Service calculates based on the IP metrics according to ITU-T G.107
Loss (max/avg,%)	The average and maximum packet loss percentage
Jitter (max/avg, ms)	The inter-arrival jitter, maximum and average, measured for the stream in microseconds
Delay (max/avg, ms)	The maximum and average round-trip delay in microseconds

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

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

26. Mitel 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 Mitel 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.

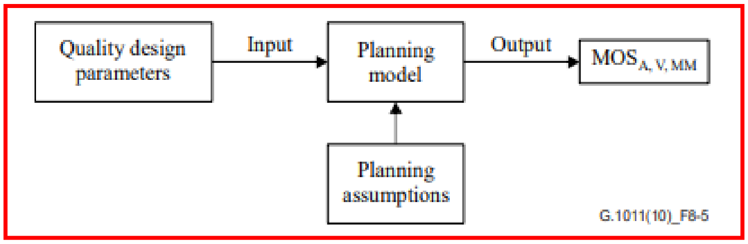


Figure 8-5 – Planning model

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)

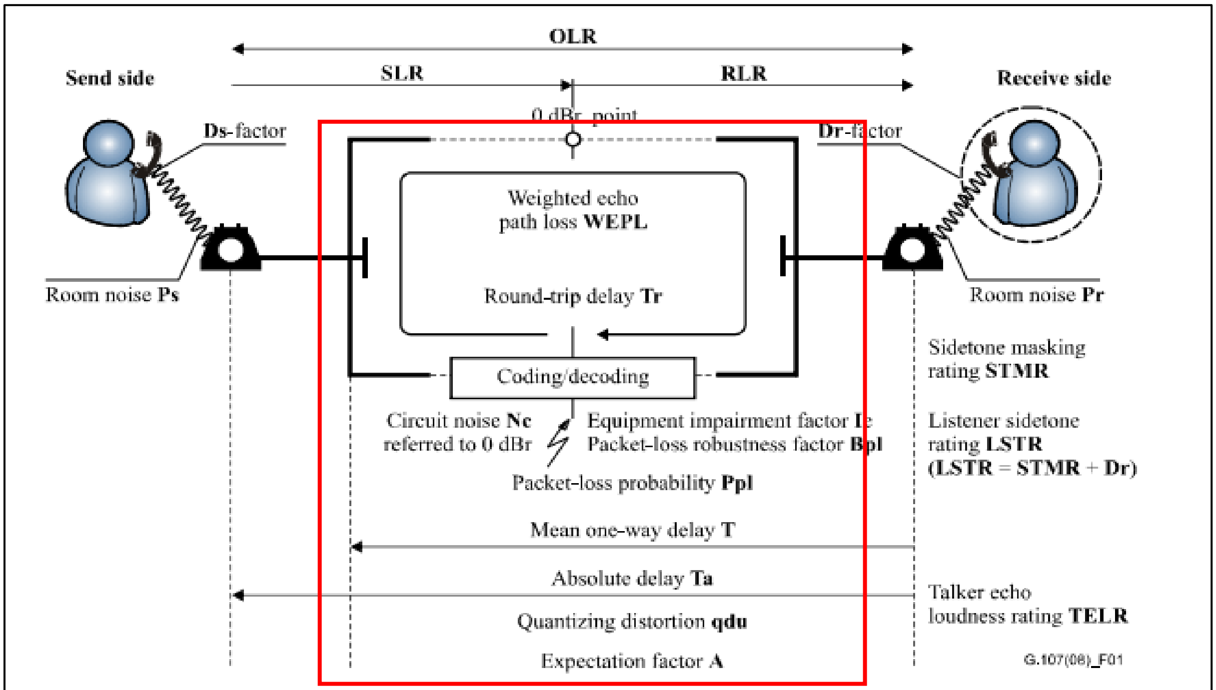


Figure 1 – Reference connection of the E-model

(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)

27. The methods practiced by Mitel'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 Mitel 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)

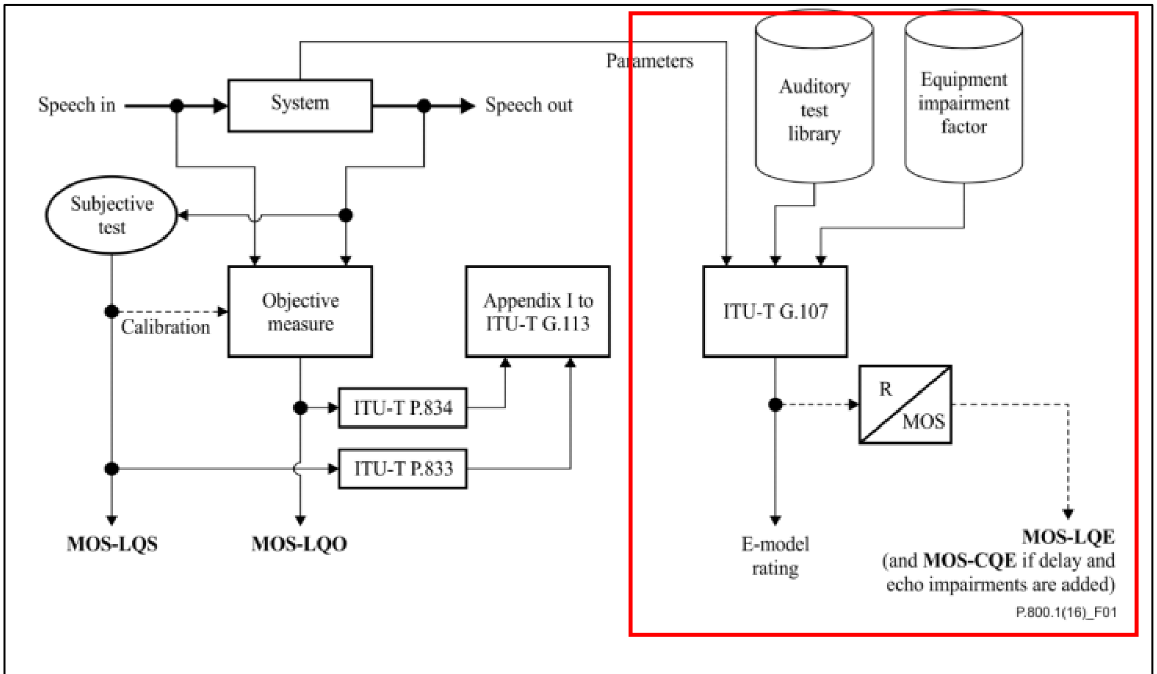


Figure 1 – Relationship between some audio MOS qualifiers

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

28. The methods practiced by Mitel'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 Mitel 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.

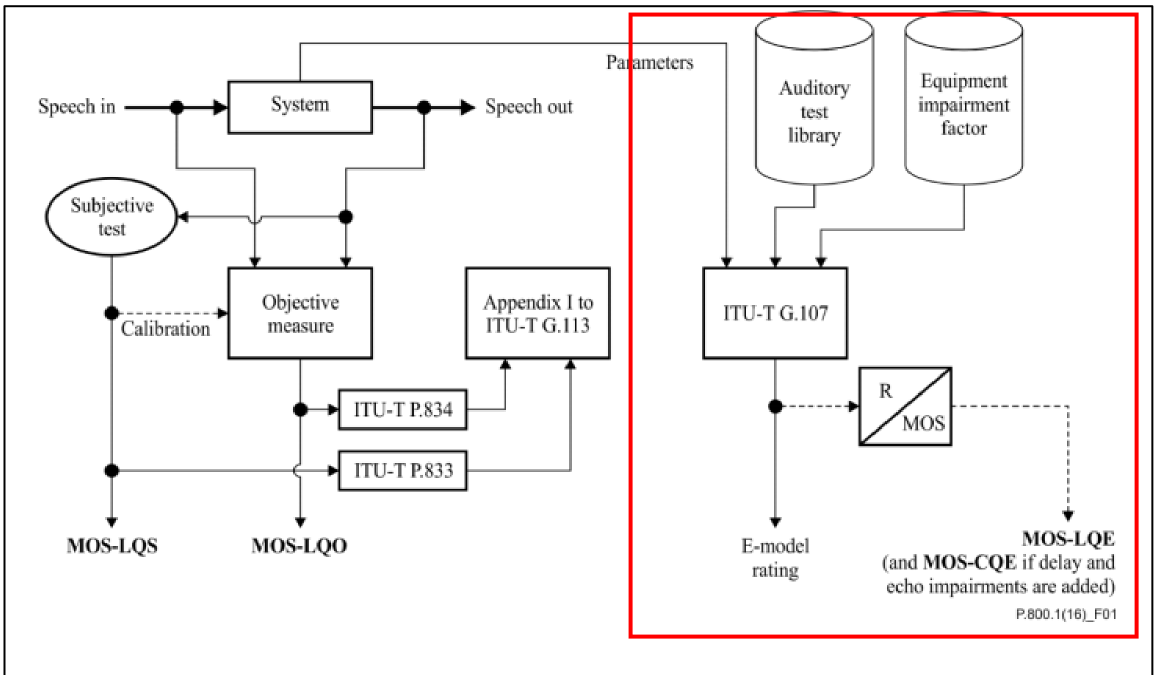


Figure 1 – Relationship between some audio MOS qualifiers

(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)

29. Mitel 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.

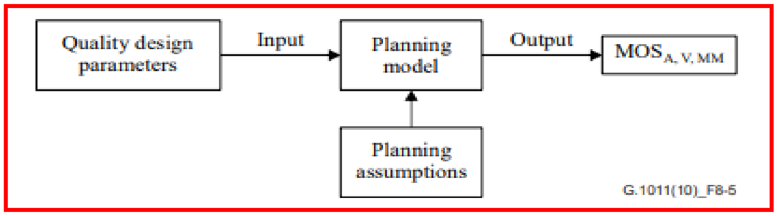


Figure 8-5 – Planning model

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=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)

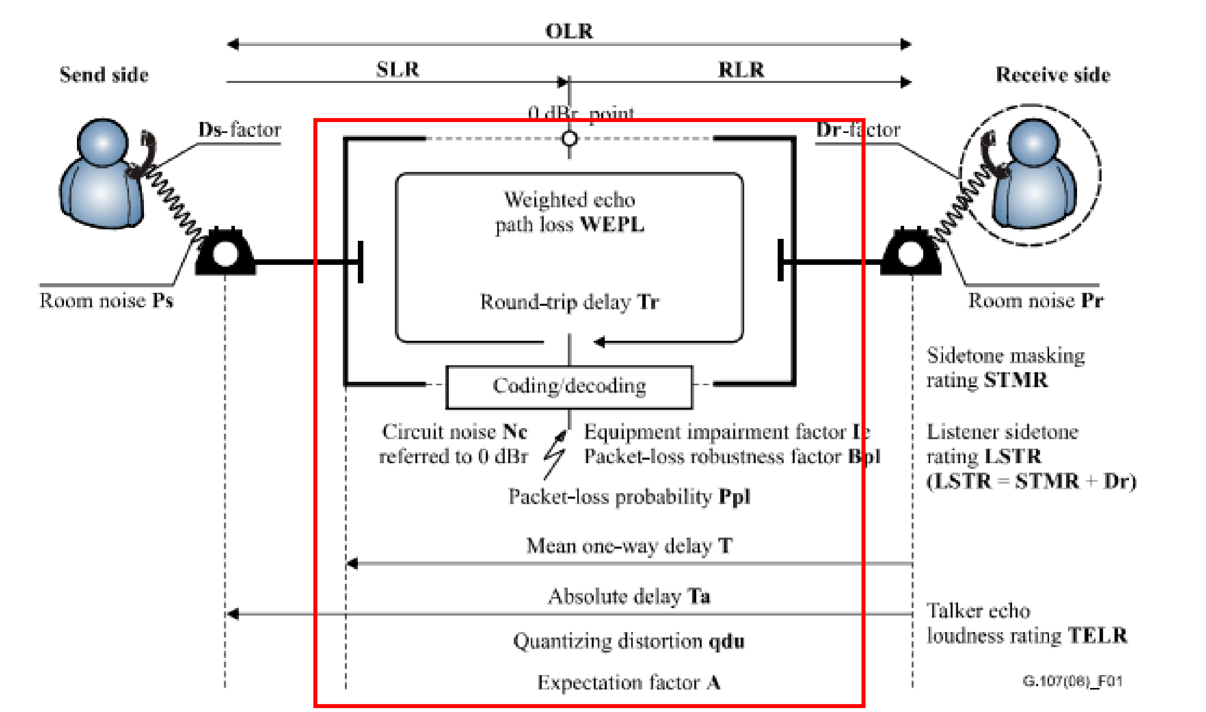


Figure 1 – Reference connection of the E-model

(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:

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

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

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)

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)

30. 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.

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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)

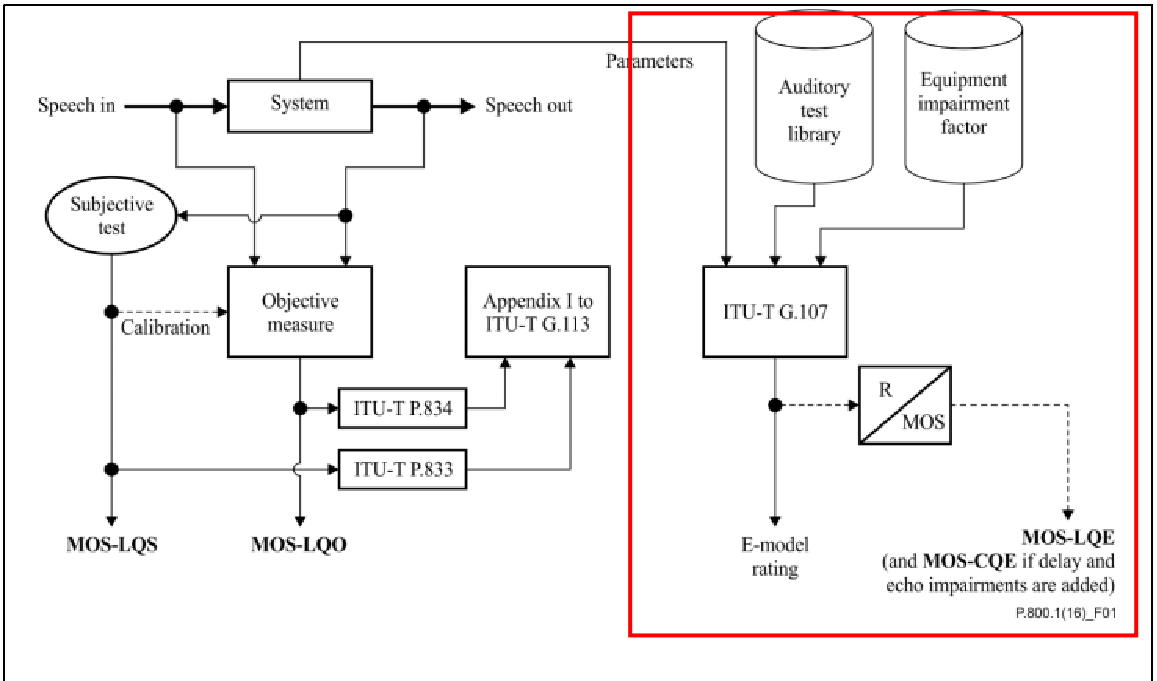

Figure 1 – Relationship between some audio MOS qualifiers

(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

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(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)

31. Mitel has had knowledge of the ‘105 Patent at least as of the date when it was notified of the filing of this action.

32. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

33. 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

34. 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."

35. 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.

36. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel Performance Analytics, Mitel Enterprise Manager, and Mitel Connect families of products that include advanced quality monitoring capabilities (collectively, "accused products").

Mitel Performance Analytics

Available with Mitel Premium Software Assurance

Key Benefits

- Faster problem detection and resolution
- Simplified management of large networks
- Improved user satisfaction and adoption
- Better use of IT resources

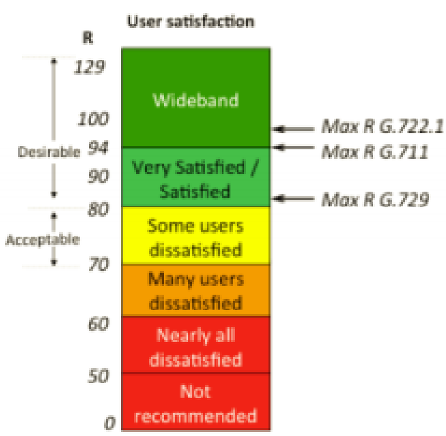
Why MPA?

- Intuitive, multi-tenant data rich dashboards
- Comprehensive testing tools
- Network diagram integration
- Reporting tools that add value

(Source : https://www.mitel.com/-/media/mitel/pdf/brochures/brochure_mpa_3_0.pdf?modified=20191016161501)

R VALUE

Mitel Performance Analytics uses the Extended E-model to estimate the received voice quality for VoIP calls. The E-model is based on ITU-T Recommendation G.107. The primary output of the E-model calculations is a scalar quality rating value known as the R value. The R scale ranges from 0 (bad) to a maximum of 129 (excellent). The range for R is 6.5 to 93.4 for narrowband codecs (G.729 and G.711) to a theoretical maximum of 129 for wideband codecs (G722.1).



(Source: https://martellotech.com/wp-content/uploads/2018/07/MPA_2.1_System_Guide.pdf)

COLOR CODING FOR VOICE QUALITY AND SIP VOICE QUALITY

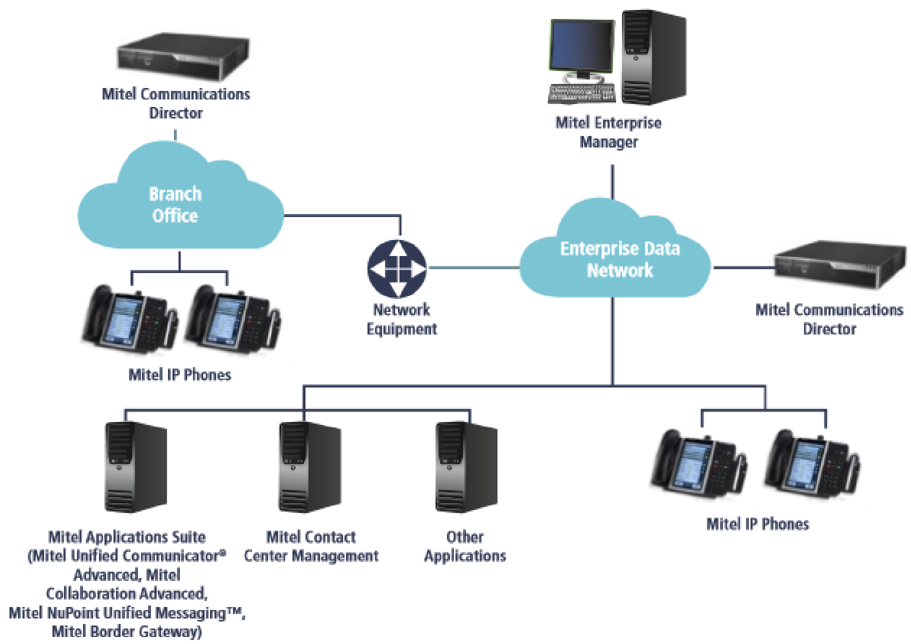
The voice quality information is color coded to enable rapid identification of trends. Mitel Performance Analytics uses the following thresholds.

		R VALUE	
COLOR	VOICE QUALITY	FROM	TO
Green	Good	80	93.2
Yellow	Adequate	70	80
Orange	Poor	60	70
Red	Bad	0	60

For good quality VoIP, the R value should be 80 or better. An R value less than 70 indicates poor audio quality and less than 60 is generally unusable.

(Source: https://martellotech.com/wp-content/uploads/2018/07/MPA_2.1_System_Guide.pdf)

BROCHURE | MITEL ENTERPRISE MANAGER



(Source :

https://www.cacommunications.com/productcatalog/mitel/pdf/MitelEnterpriseManager-Brochure.pdf)

Performance management is about optimizing system operation under normal conditions to proactively monitor and even correct potential problems before they occur. With regard to IP communications, any problems with the performance or configuration of the IP networking equipment tends to result in poor voice quality. In other words, the IP phones get the blame when in most cases poor voice quality is not the fault of the phone, but of the underlying IP network. However, the burden of proof tends to fall on the application and associated devices in which the performance problem manifests itself. This is why voice quality is the first performance management application to be supported within Mitel® Enterprise Manager.

Mitel Voice Quality Management Related Capabilities

Enterprise Manager, as well as our platform systems and associated IP phones, provides for multiple levels of voice quality monitoring and troubleshooting capabilities. Basic voice quality management tools are built into the platform systems and IP phones for monitoring and troubleshooting of solutions at an individual system level. For ongoing monitoring of voice quality over a network of multiple systems, Enterprise Manager offers the Voice Quality Manager (VQM) option. Enterprise Manager VQM provides a network-wide view of voice quality including drill-

Mitel utilizes a hierarchical approach to monitor all calls in a multi-system network. Each of the individual Mitel 3300 IP Communications Platform (ICP) systems collects statistics from their attached phones. The VQM option within Enterprise Manager collects the call quality statistics from all the 3300 ICP systems and provides a single point for integration with external management systems. For troubleshooting voice quality or other IP network problems, Enterprise Manager includes diagnostic tools for monitoring network interfaces, performing remote trace routes and

(Source :

http://www.ashtelecom.co.uk/imgpdf/92_378_Voice_Quality_Performance_Management.pdf)

Rating Voice Quality

All Mitel IP phones[2] collect and report voice quality statistics suitable for use in generating voice quality scores such as R-Value or Mean Opinion Score (MOS). Both R-Value and MOS are methods for rating user satisfaction levels for telephony voice quality. The R-Value is a quality rating factor that is the primary computational output from the ITU-T G.107 recommendation for transmission planning known as the E-Model. R-Value can be used as an input into a Mean Opinion Score (MOS) computation, that attempts to assess conversational voice quality taking into account human opinion factors associated with various types and levels of impairments.

The Enterprise Manager reporting option[3] produces voice quality reports using the R-Value scale to provide a zero to 100 quality rating with 95 being the highest rating possible for a G.711 codec-based VoIP call. Many more specialized VoIP performance management tools such as Viola NetAlly apply additional analysis to take into account human opinion factors to arrive at a MOS score. A MOS score is a subjective value from 1 (lowest) to 5 (highest) that rates the voice quality of a call and should reflect an average of a large sampling of user opinions obtained on calls with known impairments. There are many potential algorithms and variations that can be used for computing a MOS score for an IP telephony call. There is no standard MOS algorithm to use at this time and MOS scoring cannot be directly compared unless the same algorithm is used. The table below provides a general comparison between the R-value and MOS scales, though it is just an estimation.

R-Value	MOS	User Satisfaction
90-100	4.34-4.5	Very Satisfied
80-89	4.03-4.33	Satisfied
70-79	3.60-4.02	Some users dissatisfied
60-69	3.10-3.59	Many users dissatisfied
<59	<3.10	Nearly all users dissatisfied

(Source :

http://www.ashtelecom.co.uk/imgpdf/92_378_Voice_Quality_Performance_Management.pdf)

Powering connections

Mitel Connect System Administration Guide

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

Monitoring Call Quality

The Call Quality page enables easy troubleshooting of problems related to call and network quality by providing access to records of every media stream that occurs in the Mitel Connect system. The call quality metrics are derived from monitoring IP network impairments such as packet loss and delay.

Aspects of Call Quality

Call Quality is evaluated based on thresholds for packet loss, delay/latency, and jitter.

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

Table 191: Fields on the Call Quality Details Tab (Continued)

Field Name	Description
Codec	The audio codec and sample rate used in the stream If more than one codec is used for a call, only the last codec used for the call is displayed.
MOS	The stream's MOS, which the Monitoring Service calculates based on the IP metrics according to ITU-T G.107
Loss (max/avg,%)	The average and maximum packet loss percentage
Jitter (max/avg, ms)	The inter-arrival jitter, maximum and average, measured for the stream in microseconds
Delay (max/avg, ms)	The maximum and average round-trip delay in microseconds

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

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

38. Mitel 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 Mitel 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)

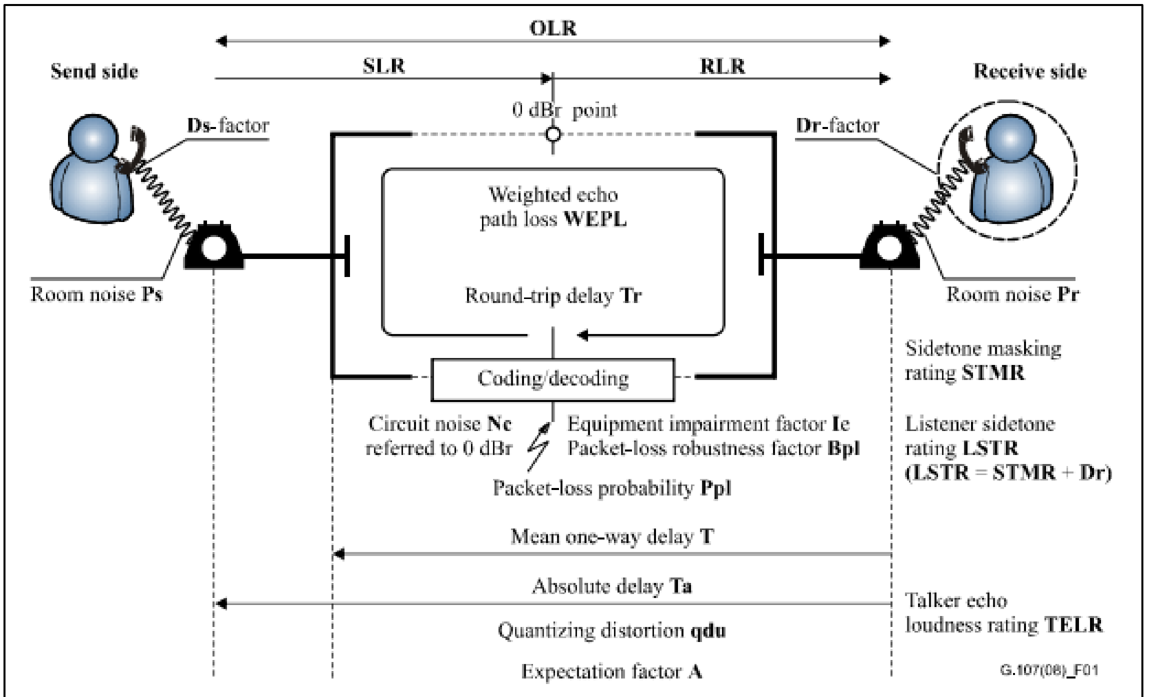


Figure 1 – Reference connection of the E-model

(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:

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

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

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:

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

(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.

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)

39. The methods practiced by Mitel'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 Mitel 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)

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

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	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 Mitel'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 Mitel 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*.

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

(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.

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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)

41. The methods practiced by Mitel'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 Mitel 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)

42. The methods practiced by Mitel'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 Mitel 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:

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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)

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

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

45. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

46. 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.

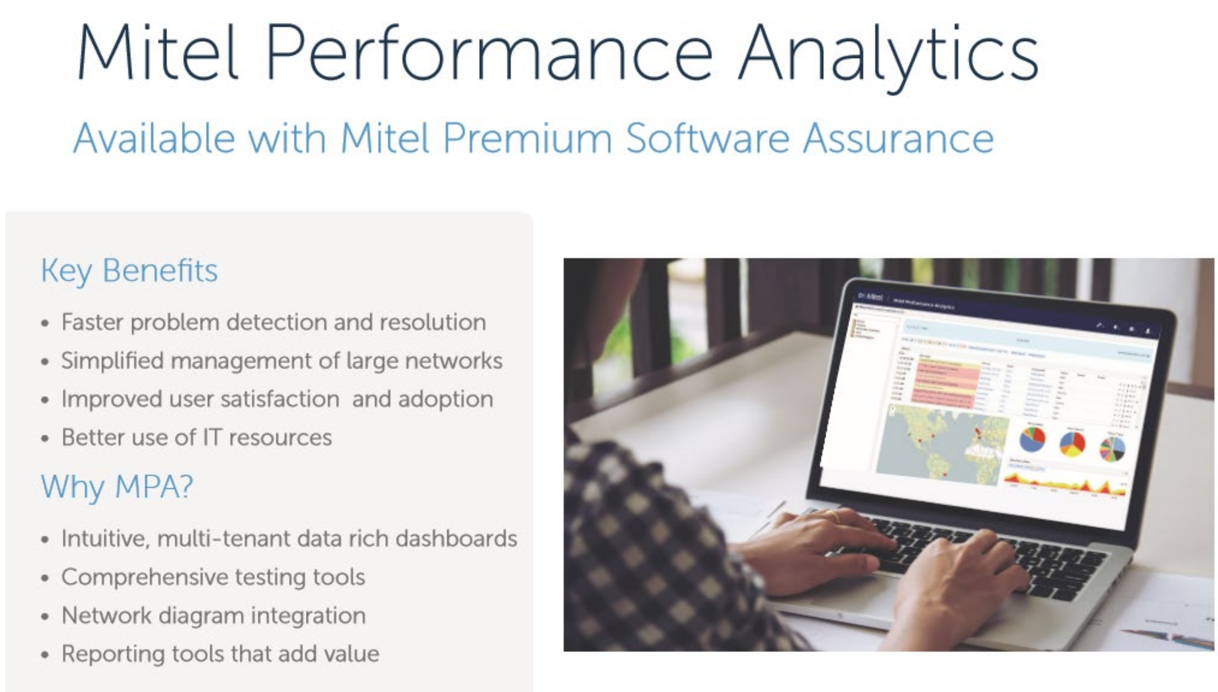
COUNT III

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

47. 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."

48. 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.

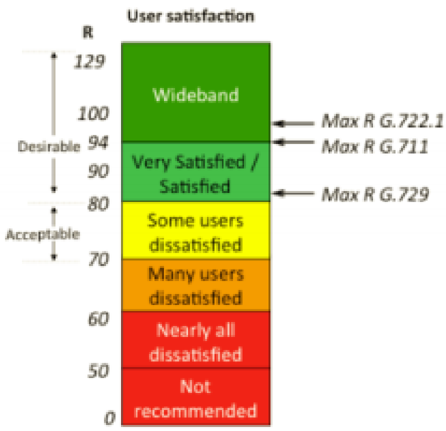
49. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel Performance Analytics, Mitel Enterprise Manager, and Mitel Connect families of products that include advanced quality monitoring capabilities (collectively, "accused products").



(Source : https://www.mitel.com/-/media/mitel/pdf/brochures/brochure_mpa_3_0.pdf?modified=20191016161501)

R VALUE

Mitel Performance Analytics uses the Extended E-model to estimate the received voice quality for VoIP calls. The E-model is based on ITU-T Recommendation G.107. The primary output of the E-model calculations is a scalar quality rating value known as the R value. The R scale ranges from 0 (bad) to a maximum of 129 (excellent). The range for R is 6.5 to 93.4 for narrowband codecs (G.729 and G.711) to a theoretical maximum of 129 for wideband codecs (G722.1).



(Source: https://martellotech.com/wp-content/uploads/2018/07/MPA_2.1_System_Guide.pdf)

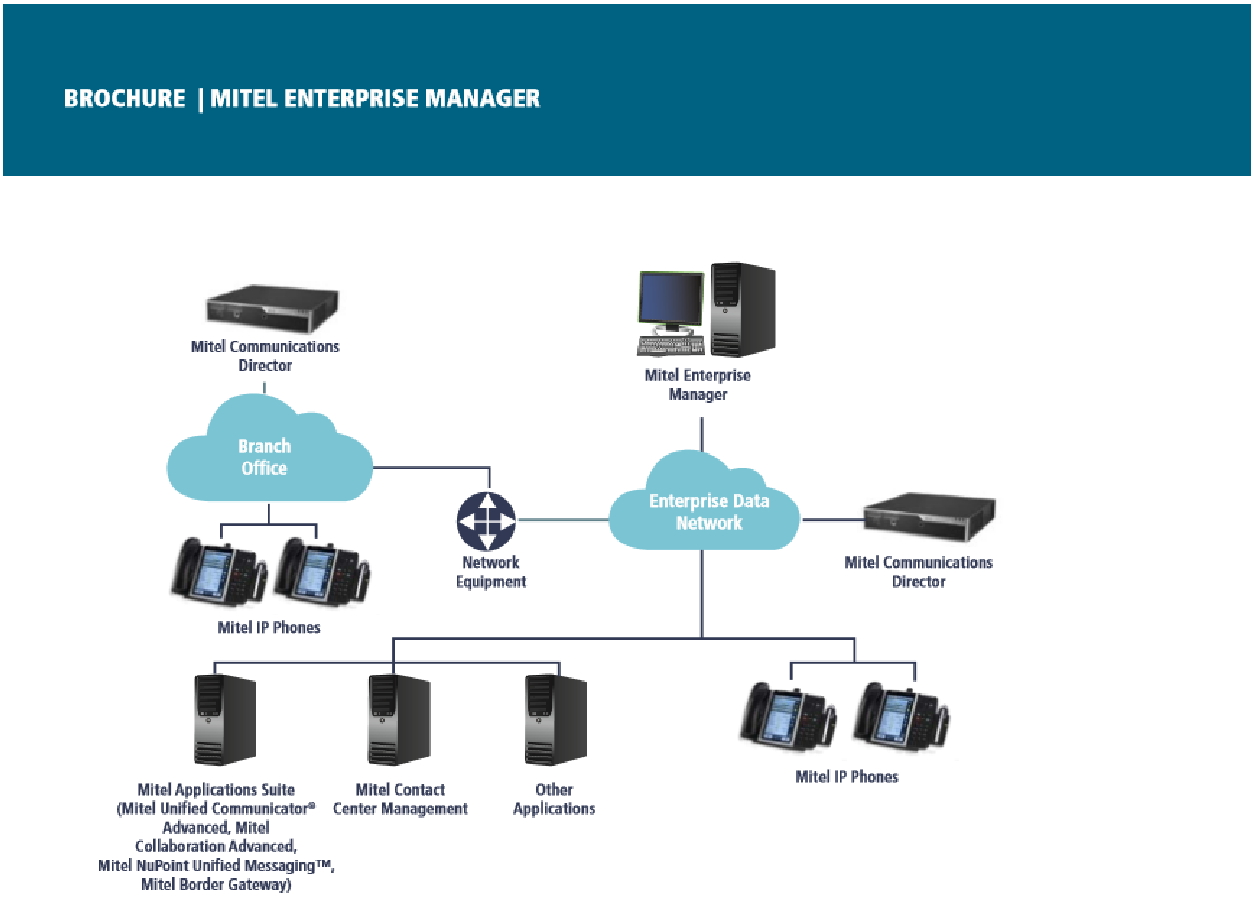
COLOR CODING FOR VOICE QUALITY AND SIP VOICE QUALITY

The voice quality information is color coded to enable rapid identification of trends. Mitel Performance Analytics uses the following thresholds.

		R VALUE	
COLOR	VOICE QUALITY	FROM	TO
Green	Good	80	93.2
Yellow	Adequate	70	80
Orange	Poor	60	70
Red	Bad	0	60

For good quality VoIP, the R value should be 80 or better. An R value less than 70 indicates poor audio quality and less than 60 is generally unusable.

(Source: https://martellotech.com/wp-content/uploads/2018/07/MPA_2.1_System_Guide.pdf)



(Source :

https://www.cacommunications.com/productcatalog/mitel/pdf/MitelEnterpriseManager-

Brochure.pdf)

Performance management is about optimizing system operation under normal conditions to proactively monitor and even correct potential problems before they occur. With regard to IP communications, any problems with the performance or configuration of the IP networking equipment tends to result in poor voice quality. In other words, the IP phones get the blame when in most cases poor voice quality is not the fault of the phone, but of the underlying IP network. However, the burden of proof tends to fall on the application and associated devices in which the performance problem manifests itself. This is why voice quality is the first performance management application to be supported within Mitel® Enterprise Manager.

Mitel Voice Quality Management Related Capabilities

Enterprise Manager, as well as our platform systems and associated IP phones, provides for multiple levels of voice quality monitoring and troubleshooting capabilities. Basic voice quality management tools are built into the platform systems and IP phones for monitoring and troubleshooting of solutions at an individual system level. For ongoing monitoring of voice quality over a network of multiple systems, Enterprise Manager offers the Voice Quality Manager (VQM) option. Enterprise Manager VQM provides a network-wide view of voice quality including drill-

Mitel utilizes a hierarchical approach to monitor all calls in a multi-system network. Each of the individual Mitel 3300 IP Communications Platform (ICP) systems collects statistics from their attached phones. The VQM option within Enterprise Manager collects the call quality statistics from all the 3300 ICP systems and provides a single point for integration with external management systems. For troubleshooting voice quality or other IP network problems, Enterprise Manager includes diagnostic tools for monitoring network interfaces, performing remote trace routes and

(Source :

http://www.ashtelecom.co.uk/imgpdf/92_378_Voice_Quality_Performance_Management.pdf)

Rating Voice Quality

All Mitel IP phones[2] collect and report voice quality statistics suitable for use in generating voice quality scores such as R-Value or Mean Opinion Score (MOS). Both R-Value and MOS are methods for rating user satisfaction levels for telephony voice quality. The R-Value is a quality rating factor that is the primary computational output from the ITU-T G.107 recommendation for transmission planning known as the E-Model. R-Value can be used as an input into a Mean Opinion Score (MOS) computation, that attempts to assess conversational voice quality taking into account human opinion factors associated with various types and levels of impairments.

The Enterprise Manager reporting option[3] produces voice quality reports using the R-Value scale to provide a zero to 100 quality rating with 95 being the highest rating possible for a G.711 codec-based VoIP call. Many more specialized VoIP performance management tools such as Viola NetAlly apply additional analysis to take into account human opinion factors to arrive at a MOS score. A MOS score is a subjective value from 1 (lowest) to 5 (highest) that rates the voice quality of a call and should reflect an average of a large sampling of user opinions obtained on calls with known impairments. There are many potential algorithms and variations that can be used for computing a MOS score for an IP telephony call. There is no standard MOS algorithm to use at this time and MOS scoring cannot be directly compared unless the same algorithm is used. The table below provides a general comparison between the R-value and MOS scales, though it is just an estimation.

R-Value	MOS	User Satisfaction
90-100	4.34-4.5	Very Satisfied
80-89	4.03-4.33	Satisfied
70-79	3.60-4.02	Some users dissatisfied
60-69	3.10-3.59	Many users dissatisfied
<59	<3.10	Nearly all users dissatisfied

(Source :

http://www.ashtelecom.co.uk/imgpdf/92_378_Voice_Quality_Performance_Management.pdf)

Powering connections

Mitel Connect System Administration Guide

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

Monitoring Call Quality

The Call Quality page enables easy troubleshooting of problems related to call and network quality by providing access to records of every media stream that occurs in the Mitel Connect system. The call quality metrics are derived from monitoring IP network impairments such as packet loss and delay.

Aspects of Call Quality

Call Quality is evaluated based on thresholds for packet loss, delay/latency, and jitter.

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

Table 191: Fields on the Call Quality Details Tab (Continued)

Field Name	Description
Codec	The audio codec and sample rate used in the stream If more than one codec is used for a call, only the last codec used for the call is displayed.
MOS	The stream's MOS, which the Monitoring Service calculates based on the IP metrics according to ITU-T G.107
Loss (max/avg,%)	The average and maximum packet loss percentage
Jitter (max/avg, ms)	The inter-arrival jitter, maximum and average, measured for the stream in microseconds
Delay (max/avg, ms)	The maximum and average round-trip delay in microseconds

(Source :

https://oneview.mitel.com/servlet/fileField?entityId=ka40h000000GtldAAC&field=Attachment_1__Body__s)

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

51. Mitel 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 Mitel 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)

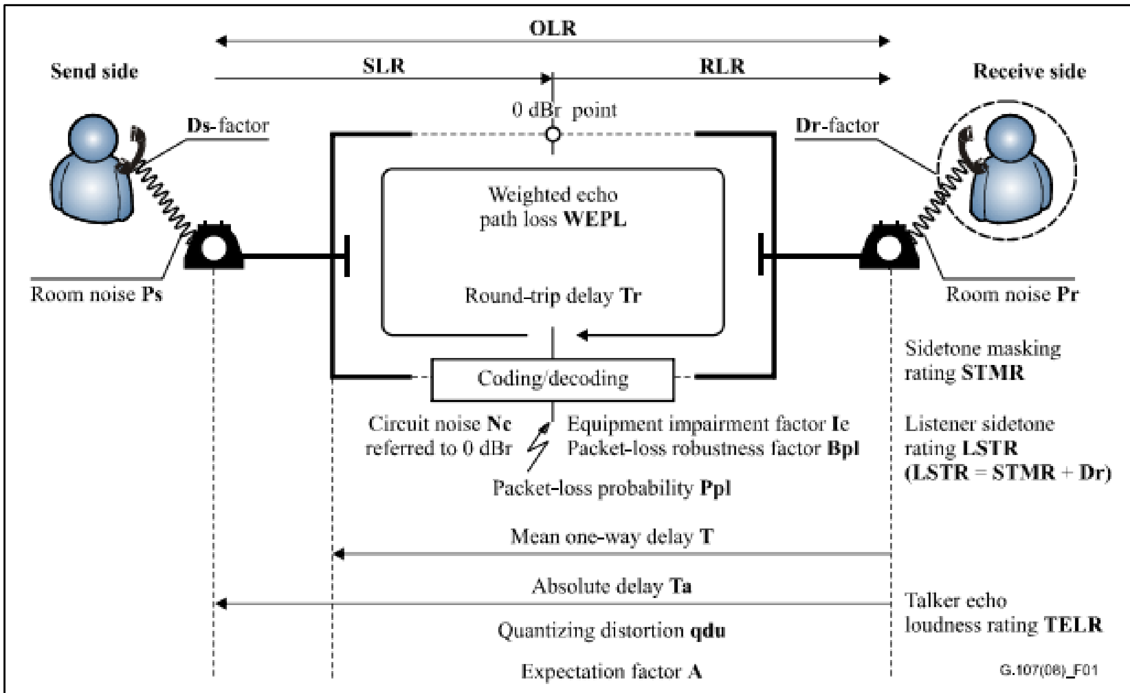


Figure 1 – Reference connection of the E-model

(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:

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

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)

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)

52. The methods practiced by Mitel'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 Mitel 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.

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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)

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

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(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)

53. The methods practiced by Mitel'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 Mitel 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 $Ie\text{-}eff$ 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)

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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)

54. The methods practiced by Mitel'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 Mitel 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

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Range of E-model Rating R	Speech transmission quality category	User satisfaction
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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)

55. Far North Patents only asserts method claims from the '230 Patent.

56. Mitel has had knowledge of the '230 Patent at least as of the date when it was notified of the filing of this action.

57. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

58. 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

59. 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."

60. 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.

61. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel TA7100 Terminal Adapter and Mitel 6900 Series IP Phones families of products that include advanced quality of service capabilities (collectively, "accused products").

MITEL TA7100 TERMINAL ADAPTERS

The Mitel TA7100 terminal adapters enable analog phones and/or faxes to connect to an IP network using SIP protocol

(Source : https://www.mitel.com/products/devices-accessories/analog-phones-peripherals/mitel-ta7100-terminal-adapters)

Networking	
Compliant Standards :	IEEE 802.1p, IEEE 802.1Q
Connectivity Technology :	Wired
Data Link Protocol :	Ethernet, Fast Ethernet

(Source: https://www.cdw.com/product/mitel-ta7108-voip-phone-adapter/5447488)

Mitel Releases the 6900 Series IP Phones

OCTOBER 28, 2016 BY CODY BLACKLEY

Mitel is adding to their business class phones by launching the 6900 series IP Phones, allowing users to take full advantage of being mobile in their workplace.

The phones allow users to pair their mobile phone and with the Bluetooth interface built into the 6900's, enabling users to handle calls from cell phones and IP calls seamlessly one one device.

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THE LINE UP:

The three versions in the Mitel 6900 series are the 6920, 6930, and the executive level 6940. The 6920 has a 3.5-inch color display screen, with 18 programmable keys, LCD display and mobile device integration through a USB dongle. The 6930 has a 4.7-inch color display screen, with 72 programmable keys, and MobileLink device integration. Mitel's 6940 is the executive level phone and it has a 7-inch touchscreen color display, Bluetooth 4.1, with 96 programmable keys, enhanced full-duplex speakerphone, and MobileLink device integration.

(Source : https://www.voipsupply.com/blog/voip-insider/mitel-releases-6900-series-ip-phone-now-one-touch-integration/)

NETWORKING

Mitel MiVoice 6920 IP Phone - VoIP phone Specs

Mitel MiVoice 6920 IP Phone - VoIP phone | 50006767

Post a comment

SPECS

MISCELLANEOUS	Color Category	black
	Placing / Mounting	table-top, wall-mountable
IP TELEPHONY	VoIP Protocols	MiNet
	Voice Codecs	G.711a, G.711u, G.722, G.722.1, G.729a
	Quality of Service	IEEE 802.1Q (VLAN), IEEE 802.1p
	VoIP	Yes

(Source : https://www.cnet.com/products/mitel-mivoice-6920-ip-phone-voip-phone/)

Overview Tech Specs

Main Features

- VoIP phone
- Bluetooth interface
- MiNet
- multiline

The MiVoice 6930 is designed for power users who need a phone that can be tailored to their specific communication needs. The 6930 affords users the flexibility to tailor the phone for specific needs. It's designed from the ground up to provide an exceptional audio experience via its unique speech optimized handset, enhanced full-duplex speakerphone and support for Bluetooth, USB and analog headsets. Supporting high speed networks through dual Gigabit Ethernet ports, the 6930 offers a large color backlit LCD display, wideband audio with advanced audio processing, programmable personal keys and context sensitive soft keys.

Tech Specs

Specifications are provided by the manufacturer.

IP Telephony	
Compatible Software :	MiVoice Business, MiVoice Office
Lines Supported :	Multiline
Main Features :	Integrated Ethernet switch
Network Features :	Class 3 PoE
Network Ports Qty :	2 x Ethernet 10Base-T/100Base-TX/1000Base-T
Power Over Ethernet (PoE) Support :	Yes
Quality of Service :	IEEE 802.1p, IEEE 802.1Q (VLAN)

(Source : https://www.cdw.com/product/Mitel-MiVoice-6930-IP-Phone-VoIP-phone-Bluetooth-interface/4394539)

Overview Tech Specs Accessories

Main Features

- VoIP phone
- Bluetooth interface
- MiNet

Designed for the executive user who demands an exceptional phone that meets their demanding communication needs, the MiVoice 6940 IP Phone offers power users a touch-centric user experience on top of robust, productivity-enhancing features.

Mobile device integration seamlessly marries your mobile phone call audio and contact information with the MiVoice 6940. Calls to your mobile phone can be answered on the MiVoice 6940 just like any other call, leveraging the superior audio performance and ergonomics of the 6940 IP Phone. Mobile phone contacts are automatically synchronized with the MiVoice 6940 allowing access to the same contacts on either device. Additionally a powered USB port suitable for charging a mobile phone is also built into the MiVoice 6940.

Tech Specs

Specifications are provided by the manufacturer.

IP Telephony	
Compatible Software :	MiVoice Business, MiVoice Office
Main Features :	Integrated Ethernet switch
Network Features :	Class 3 PoE
Network Ports Qty :	2 x Ethernet 10Base-T/100Base-TX/1000Base-T
Power Over Ethernet (PoE) Support :	Yes
Quality of Service :	IEEE 802.1p, IEEE 802.1Q (VLAN)

(Source : https://www.cdw.com/product/mitel-mivoice-6940-ip-phone-voip-phone-bluetooth-interface/4884989)

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

63. Mitel 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)

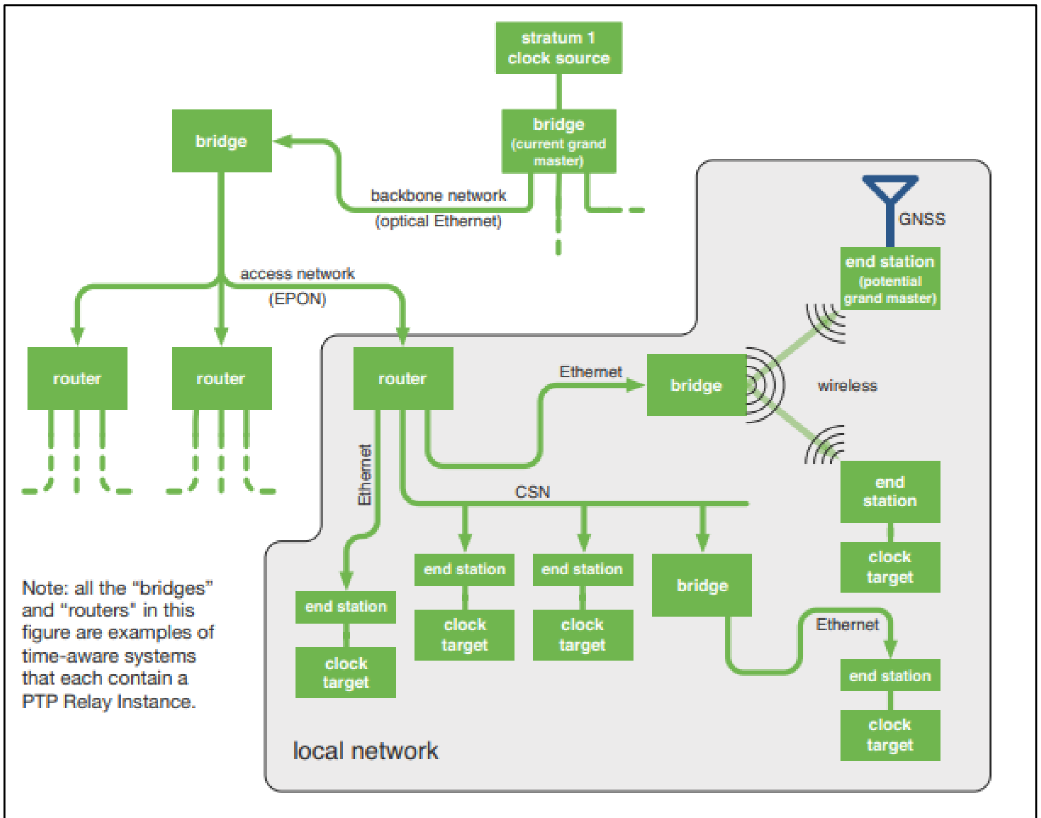

Figure 7-1—Time-aware network example

(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]

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)

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)

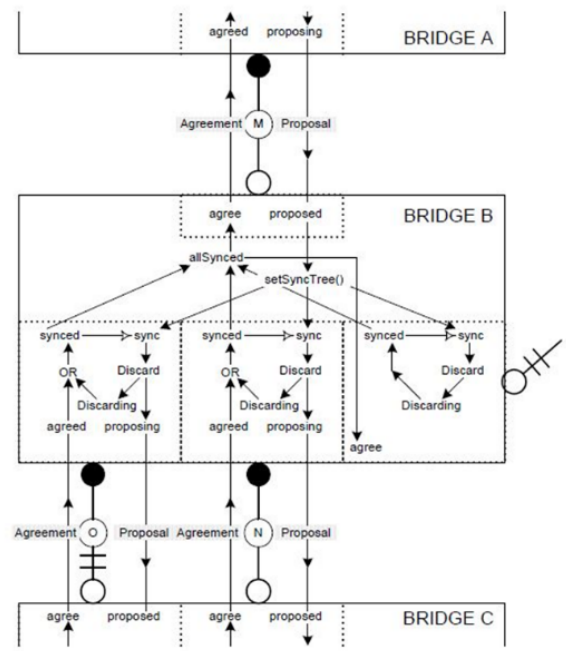

Figure 13-10—Agreements and Proposals

(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)

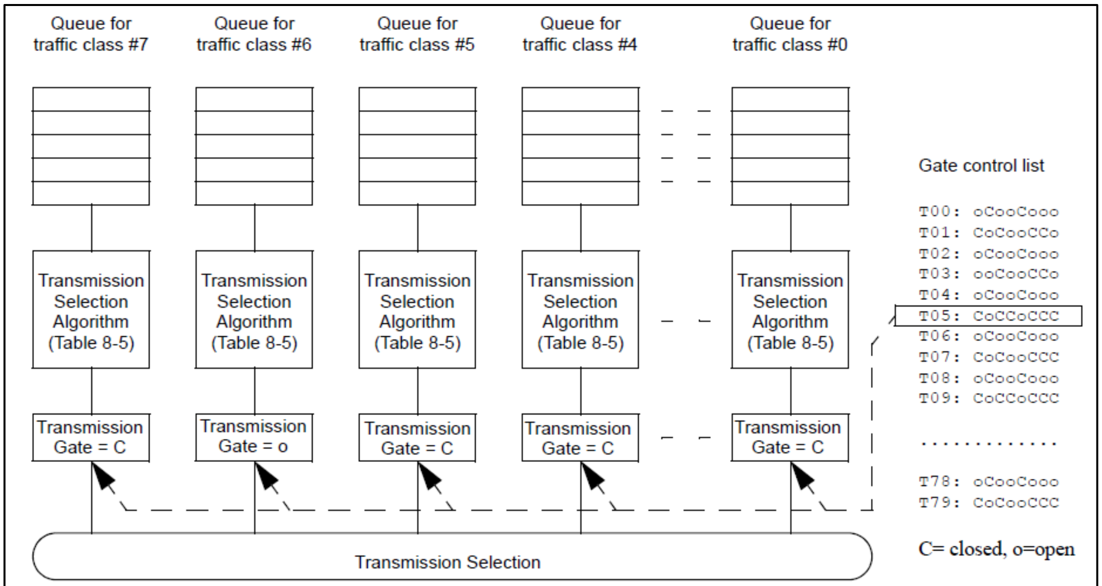

Figure 8-12—Transmission selection with gates

(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)

64. 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))

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)

65. Mitel 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 Mitel 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)

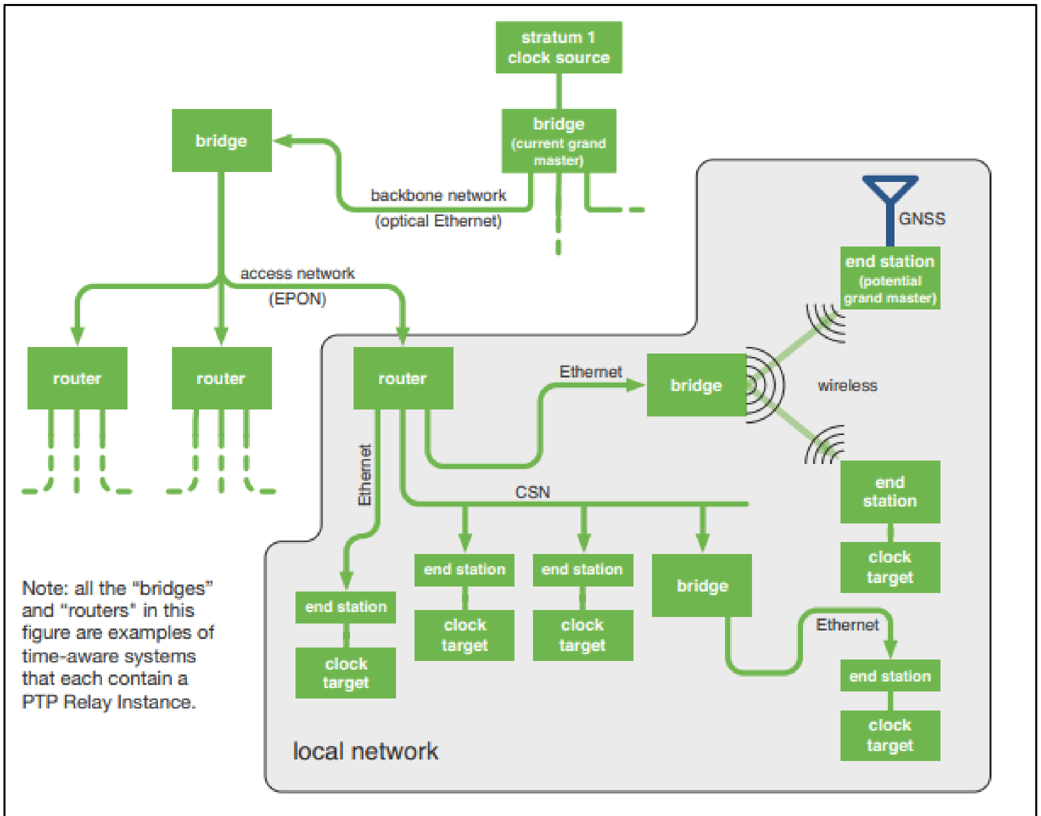

Figure 7-1—Time-aware network example

(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,

(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

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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 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))

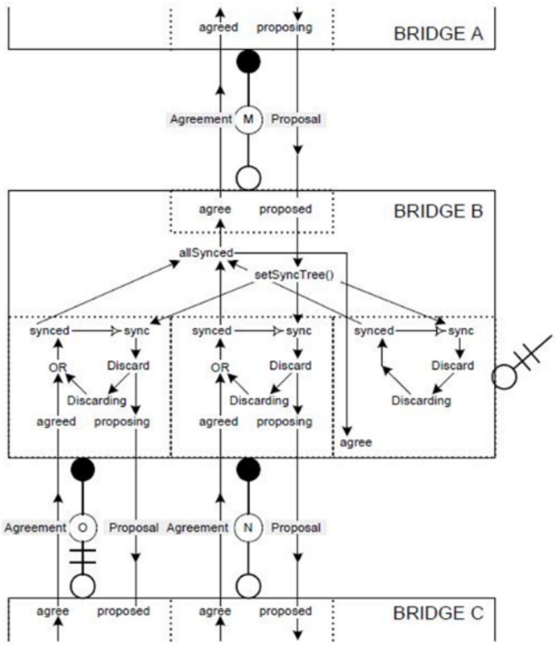

Figure 13-10—Agreements and Proposals

(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)

66. The methods practiced by Mitel'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 Mitel 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)

- **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
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    - MAC address is tie breaker
  - Time relays drop all inferior ANNOUNCE messages
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    - 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)

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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)

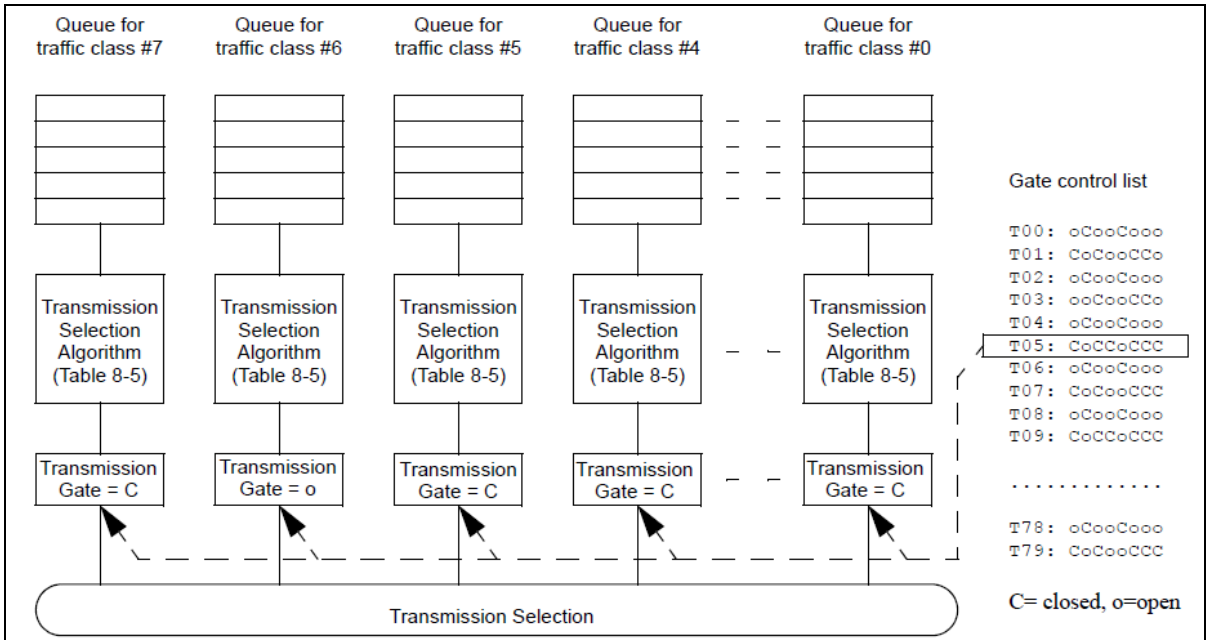

Figure 8-12—Transmission selection with gates

(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)

67. The methods practiced by Mitel'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 Mitel 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)

68. The plurality of device adaptors recited above in connection with Mitel'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 Mitel 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)

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

70. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

71. 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

72. 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."

73. 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.

74. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel TA7100 Terminal Adapter and Mitel 6900 Series IP Phones families of products that include advanced quality of service capabilities (collectively, "accused products").

MITEL TA7100 TERMINAL ADAPTERS

The Mitel TA7100 terminal adapters enable analog phones and/or faxes to connect to an IP network using SIP protocol

(Source : https://www.mitel.com/products/devices-accessories/analog-phones-peripherals/mitel-ta7100-terminal-adapters)

Networking	
Compliant Standards :	IEEE 802.1p, IEEE 802.1Q
Connectivity Technology :	Wired
Data Link Protocol :	Ethernet, Fast Ethernet

(Source: https://www.cdw.com/product/mitel-ta7108-voip-phone-adapter/5447488)

Mitel Releases the 6900 Series IP Phones

OCTOBER 28, 2016 BY CODY BLACKLEY

Mitel is adding to their business class phones by launching the 6900 series IP Phones, allowing users to take full advantage of being mobile in their workplace.

The phones allow users to pair their mobile phone and with the Bluetooth interface built into the 6900's, enabling users to handle calls from cell phones and IP calls seamlessly one one device.

The phones allow users to pair their mobile phone and with the Bluetooth interface built into the 6900's, enabling users to handle calls from cell phones and IP calls seamlessly on one device.

THE LINE UP:

The three versions in the Mitel 6900 series are the 6920, 6930, and the executive level 6940. The 6920 has a 3.5-inch color display screen, with 18 programmable keys, LCD display and mobile device integration through a USB dongle. The 6930 has a 4.7-inch color display screen, with 72 programmable keys, and MobileLink device integration. Mitel's 6940 is the executive level phone and it has a 7-inch touchscreen color display, Bluetooth 4.1, with 96 programmable keys, enhanced full-duplex speakerphone, and MobileLink device integration.

(Source : https://www.voipsupply.com/blog/voip-insider/mitel-releases-6900-series-ip-phone-now-one-touch-integration/)

NETWORKING

Mitel MiVoice 6920 IP Phone - VoIP phone Specs

Mitel MiVoice 6920 IP Phone - VoIP phone | 50006767

Post a comment

SPECS

MISCELLANEOUS	Color Category	black
	Placing / Mounting	table-top, wall-mountable
IP TELEPHONY	VoIP Protocols	MiNet
	Voice Codecs	G.711a, G.711u, G.722, G.722.1, G.729a
	Quality of Service	IEEE 802.1Q (VLAN), IEEE 802.1p
	VoIP	Yes

(Source : https://www.cnet.com/products/mitel-mivoice-6920-ip-phone-voip-phone/)

Overview Tech Specs

Main Features

- VoIP phone
- Bluetooth interface
- MiNet
- multiline

The MiVoice 6930 is designed for power users who need a phone that can be tailored to their specific communication needs. The 6930 affords users the flexibility to tailor the phone for specific needs. It's designed from the ground up to provide an exceptional audio experience via its unique speech optimized handset, enhanced full-duplex speakerphone and support for Bluetooth, USB and analog headsets. Supporting high speed networks through dual Gigabit Ethernet ports, the 6930 offers a large color backlit LCD display, wideband audio with advanced audio processing, programmable personal keys and context sensitive soft keys.

Tech Specs

Specifications are provided by the manufacturer.

IP Telephony	
Compatible Software :	MiVoice Business, MiVoice Office
Lines Supported :	Multiline
Main Features :	Integrated Ethernet switch
Network Features :	Class 3 PoE
Network Ports Qty :	2 x Ethernet 10Base-T/100Base-TX/1000Base-T
Power Over Ethernet (PoE) Support :	Yes
Quality of Service :	IEEE 802.1p, IEEE 802.1Q (VLAN)

(Source : https://www.cdw.com/product/Mitel-MiVoice-6930-IP-Phone-VoIP-phone-Bluetooth-interface/4394539)

Overview Tech Specs Accessories

Main Features

- VoIP phone
- Bluetooth interface
- MiNet

Designed for the executive user who demands an exceptional phone that meets their demanding communication needs, the MiVoice 6940 IP Phone offers power users a touch-centric user experience on top of robust, productivity-enhancing features.

Mobile device integration seamlessly marries your mobile phone call audio and contact information with the MiVoice 6940. Calls to your mobile phone can be answered on the MiVoice 6940 just like any other call, leveraging the superior audio performance and ergonomics of the 6940 IP Phone. Mobile phone contacts are automatically synchronized with the MiVoice 6940 allowing access to the same contacts on either device. Additionally a powered USB port suitable for charging a mobile phone is also built into the MiVoice 6940.

Tech Specs

Specifications are provided by the manufacturer.

IP Telephony	
Compatible Software :	MiVoice Business, MiVoice Office
Main Features :	Integrated Ethernet switch
Network Features :	Class 3 PoE
Network Ports Qty :	2 x Ethernet 10Base-T/100Base-TX/1000Base-T
Power Over Ethernet (PoE) Support :	Yes
Quality of Service :	IEEE 802.1p, IEEE 802.1Q (VLAN)

(Source : https://www.cdw.com/product/mitel-mivoice-6940-ip-phone-voip-phone-bluetooth-interface/4884989)

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

76. Mitel 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 Mitel 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)

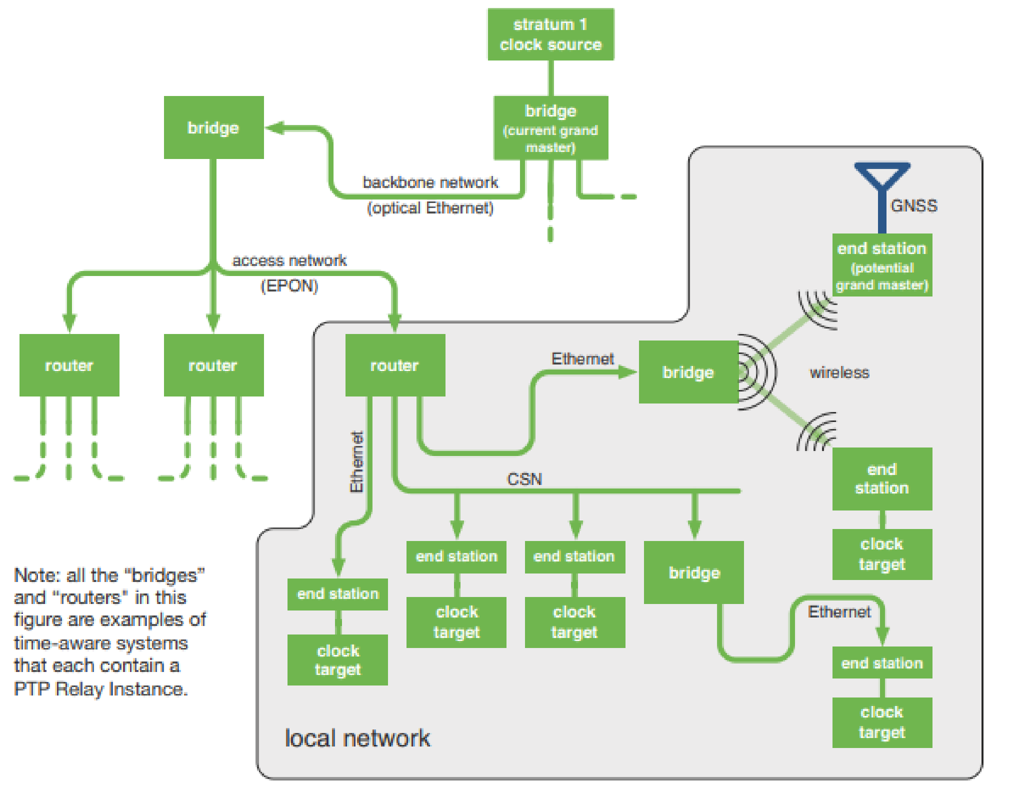

Figure 7-1—Time-aware network example

(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)

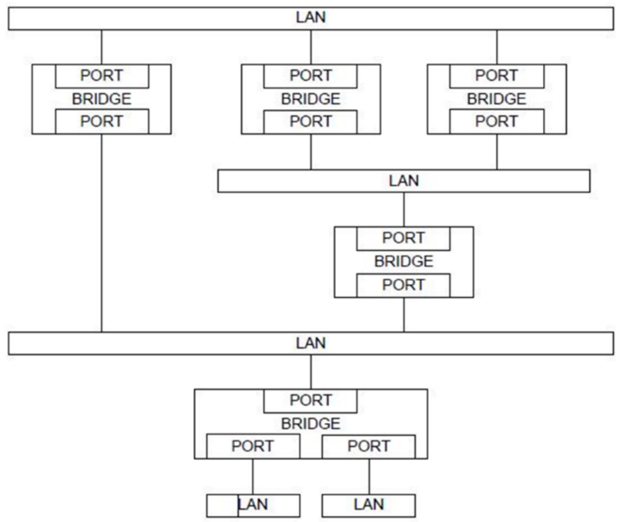

Figure 8-1—A Bridged Network

(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)

77. The methods practiced by Mitel'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 Mitel 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)

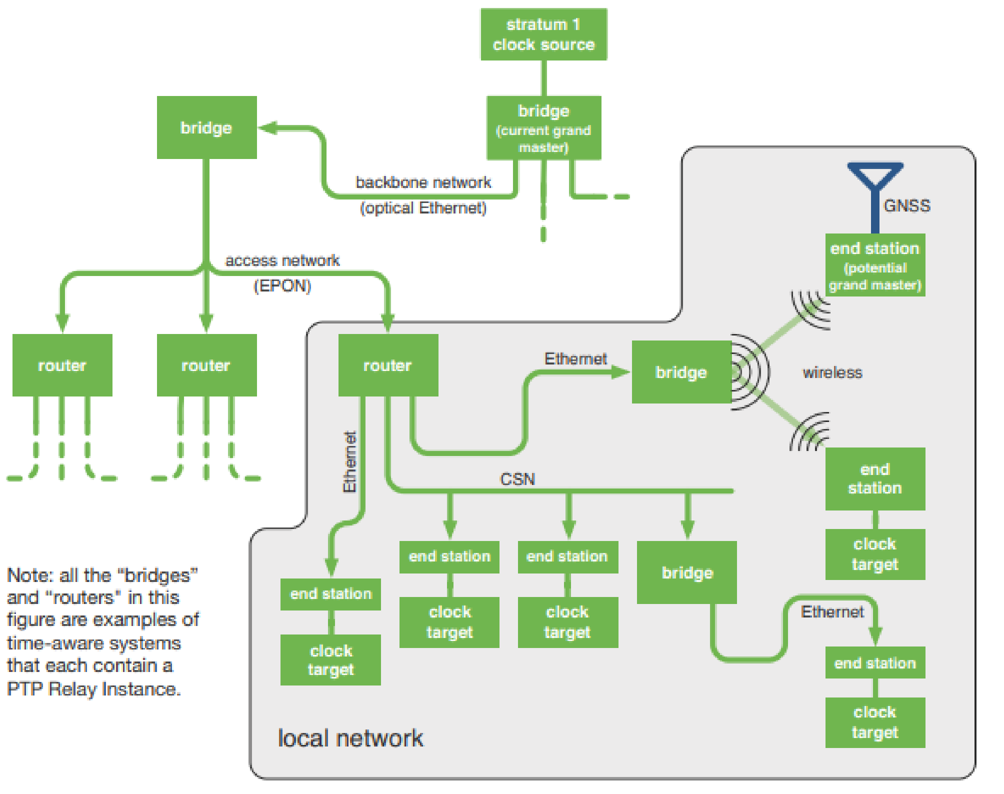

Figure 7-1—Time-aware network example

(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)

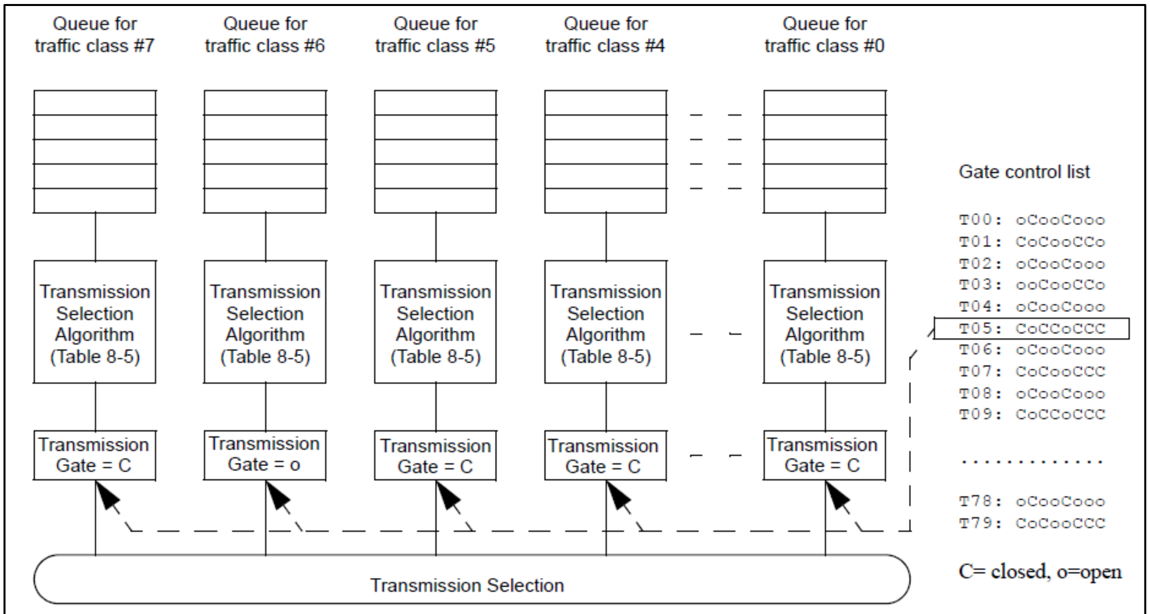

Figure 8-12—Transmission selection with gates

(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))

78. The methods practiced by Mitel'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 Mitel 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))

79. The methods practiced by Mitel'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 Mitel 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).

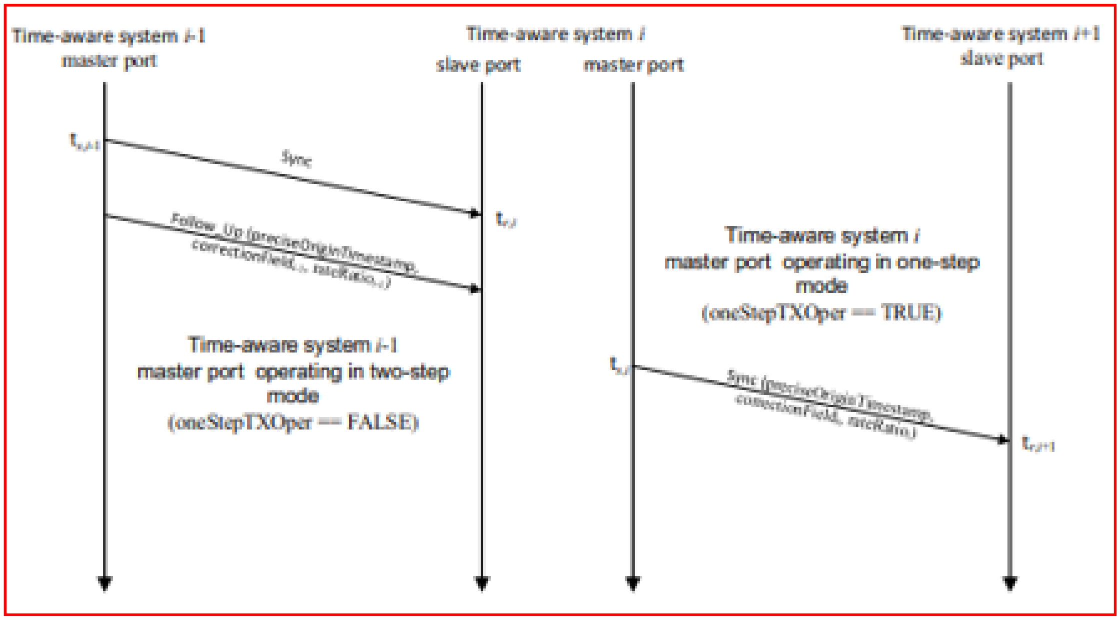

Figure 11-2—Transport of time-synchronization information

(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)

80. The methods practiced by Mitel'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 Mitel 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

(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)

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).

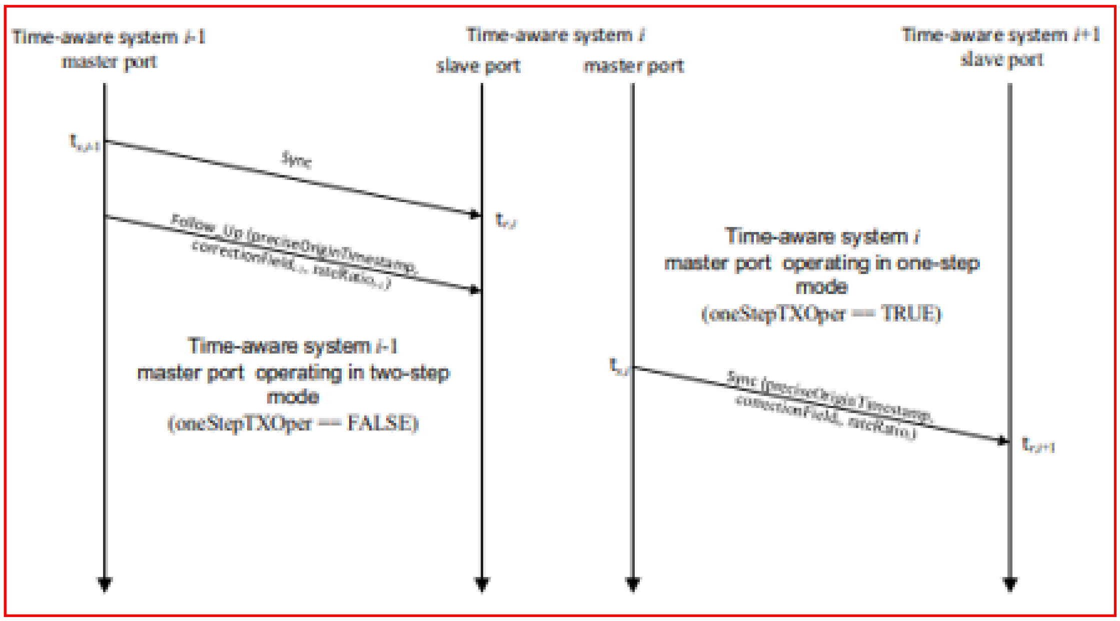

Figure 11-2—Transport of time-synchronization information

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

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

82. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

83. 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

84. 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.”

85. 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.

86. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel TA7100 Terminal Adapter and Mitel 6900 Series IP Phones families of products that include advanced quality of service capabilities (collectively, “accused products”).

MITEL TA7100 TERMINAL ADAPTERS

The Mitel TA7100 terminal adapters enable analog phones and/or faxes to connect to an IP network using SIP protocol

(Source : https://www.mitel.com/products/devices-accessories/analog-phones-peripherals/mitel-ta7100-terminal-adapters)

Mitel Releases the 6900 Series IP Phones

OCTOBER 28, 2016 BY CODY BLACKLEY

Mitel is adding to their business class phones by launching the 6900 series IP Phones, allowing users to take full advantage of being mobile in their workplace.

The phones allow users to pair their mobile phone and with the Bluetooth interface built into the 6900's, enabling users to handle calls from cell phones and IP calls seamlessly one one device.

The phones allow users to pair their mobile phone and with the Bluetooth interface built into the 6900's, enabling users to handle calls from cell phones and IP calls seamlessly on one device.

THE LINE UP:

The three versions in the Mitel 6900 series are the 6920, 6930, and the executive level 6940. The 6920 has a 3.5-inch color display screen, with 18 programmable keys, LCD display and mobile device integration through a USB dongle. The 6930 has a 4.7-inch color display screen, with 72 programmable keys, and MobileLink device integration. Mitel's 6940 is the executive level phone and it has a 7-inch touchscreen color display, Bluetooth 4.1, with 96 programmable keys, enhanced full-duplex speakerphone, and MobileLink device integration.

(Source: https://www.cdw.com/product/mitel-ta7108-voip-phone-adapter/5447488)

Mitel Releases the 6900 Series IP Phones

OCTOBER 28, 2016 BY CODY BLACKLEY

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The phones allow users to pair their mobile phone and with the Bluetooth interface built into the 6900's, enabling users to handle calls from cell phones and IP calls seamlessly on one device.

THE LINE UP:

The three versions in the Mitel 6900 series are the 6920, 6930, and the executive level 6940. The 6920 has a 3.5-inch color display screen, with 18 programmable keys, LCD display and mobile device integration through a USB dongle. The 6930 has a 4.7-inch color display screen, with 72 programmable keys, and MobileLink device integration. Mitel's 6940 is the executive level phone and it has a 7-inch touchscreen color display, Bluetooth 4.1, with 96 programmable keys, enhanced full-duplex speakerphone, and MobileLink device integration.

(Source : https://www.voipsupply.com/blog/voip-insider/mitel-releases-6900-series-ip-phone-now-one-touch-integration/)

NETWORKING

Mitel MiVoice 6920 IP Phone - VoIP phone Specs

Mitel MiVoice 6920 IP Phone - VoIP phone | 50006767

Post a comment

SPECS

MISCELLANEOUS	Color Category	black
	Placing / Mounting	table-top, wall-mountable
IP TELEPHONY	VoIP Protocols	MiNet
	Voice Codecs	G.711a, G.711u, G.722, G.722.1, G.729a
	Quality of Service	IEEE 802.1Q (VLAN), IEEE 802.1p
	VoIP	Yes

(Source : https://www.cnet.com/products/mitel-mivoice-6920-ip-phone-voip-phone/)

Overview   Tech Specs

Main Features

- VoIP phone
- Bluetooth interface
- MiNet
- multiline

The MiVoice 6930 is designed for power users who need a phone that can be tailored to their specific communication needs. The 6930 affords users the flexibility to tailor the phone for specific needs. It s designed from the ground up to provide an exceptional audio experience via its unique speech optimized handset, enhanced full-duplex speakerphone and support for Bluetooth, USB and analog headsets. Supporting high speed networks through dual Gigabit Ethernet ports, the 6930 offers a large color backlit LCD display, wideband audio with advanced audio processing, programmable personal keys and context sensitive soft keys.

Tech Specs

Specifications are provided by the manufacturer.

IP Telephony	
Compatible Software :	MiVoice Business, MiVoice Office
Lines Supported :	Multiline
Main Features :	Integrated Ethernet switch
Network Features :	Class 3 PoE
Network Ports Qty :	2 x Ethernet 10Base-T/100Base-TX/1000Base-T
Power Over Ethernet (PoE) Support :	Yes
Quality of Service :	IEEE 802.1p, IEEE 802.1Q (VLAN)

(Source : https://www.cdw.com/product/Mitel-MiVoice-6930-IP-Phone-VoIP-phone-Bluetooth-interface/4394539)

Overview   Tech Specs   Accessories

Main Features

- VoIP phone
- Bluetooth interface
- MiNet

Designed for the executive user who demands an exceptional phone that meets their demanding communication needs, the MiVoice 6940 IP Phone offers power users a touch-centric user experience on top of robust, productivity-enhancing features.

Mobile device integration seamlessly marries your mobile phone call audio and contact information with the MiVoice 6940. Calls to your mobile phone can be answered on the MiVoice 6940 just like any other call, leveraging the superior audio performance and ergonomics of the 6940 IP Phone. Mobile phone contacts are automatically synchronized with the MiVoice 6940 allowing access to the same contacts on either device. Additionally a powered USB port suitable for charging a mobile phone is also built into the MiVoice 6940.

Tech Specs

Specifications are provided by the manufacturer.

IP Telephony	
Compatible Software :	MiVoice Business, MiVoice Office
Main Features :	Integrated Ethernet switch
Network Features :	Class 3 PoE
Network Ports Qty :	2 x Ethernet 10Base-T/100Base-TX/1000Base-T
Power Over Ethernet (PoE) Support :	Yes
Quality of Service :	IEEE 802.1p, IEEE 802.1Q (VLAN)

(Source : https://www.cdw.com/product/mitel-mivoice-6940-ip-phone-voip-phone-bluetooth-interface/4884989)

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

88. Mitel 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 Mitel 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).

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

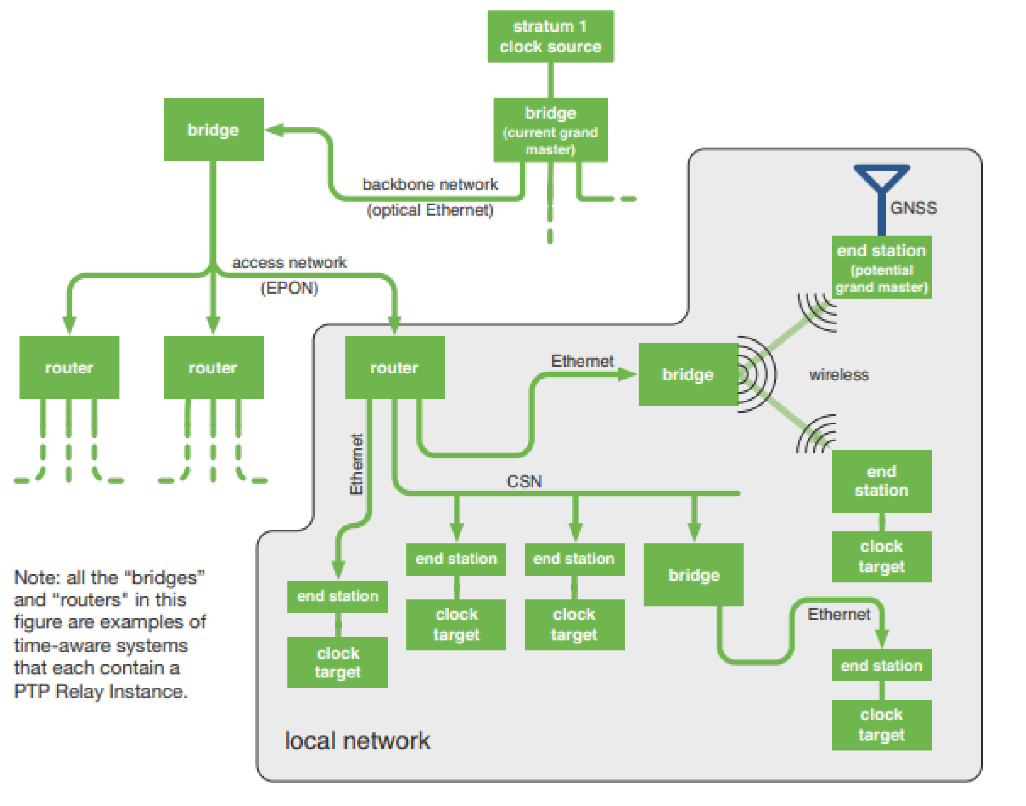


Figure 7-1—Time-aware network example

(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)

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)

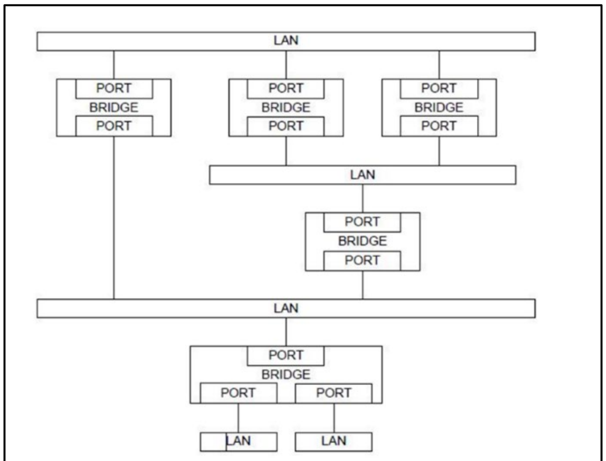


Figure 8-1—A Bridged Network

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

89. The methods practiced by Mitel'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 Mitel 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

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

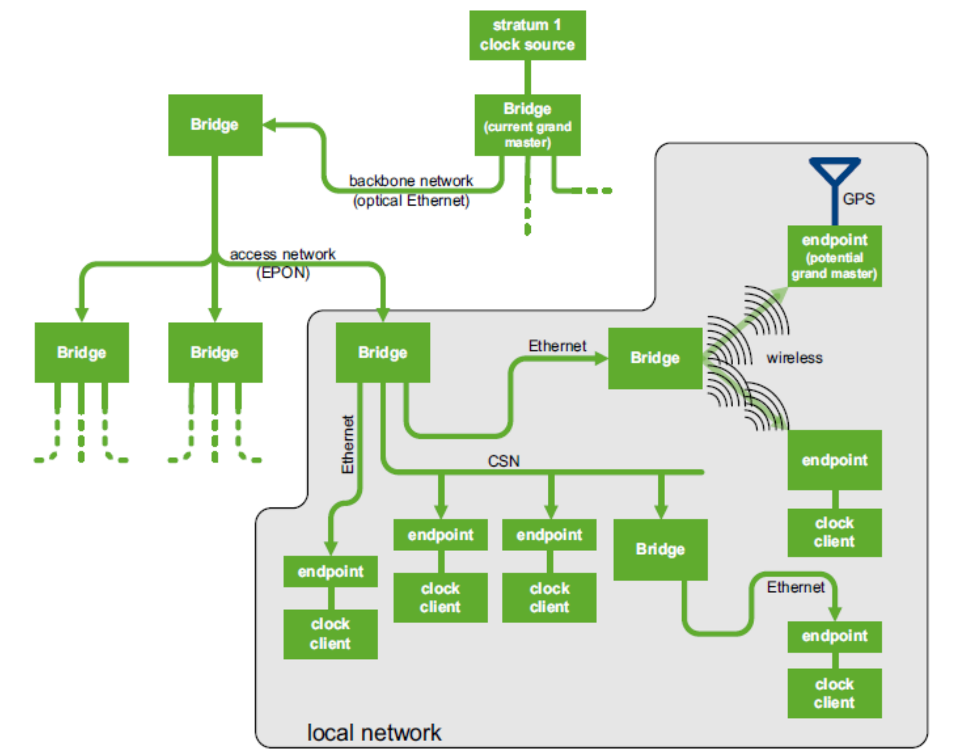


Figure 7-1—Time-aware network example

(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)

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)

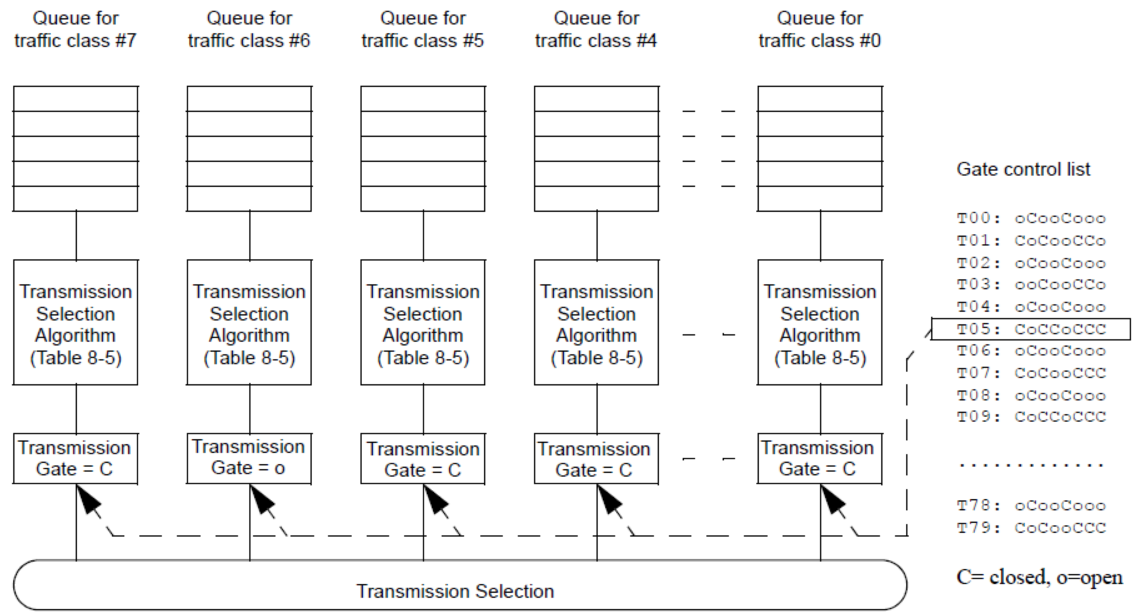


Figure 8-12—Transmission selection with gates

(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))

90. The methods practiced by Mitel'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 Mitel 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))

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).

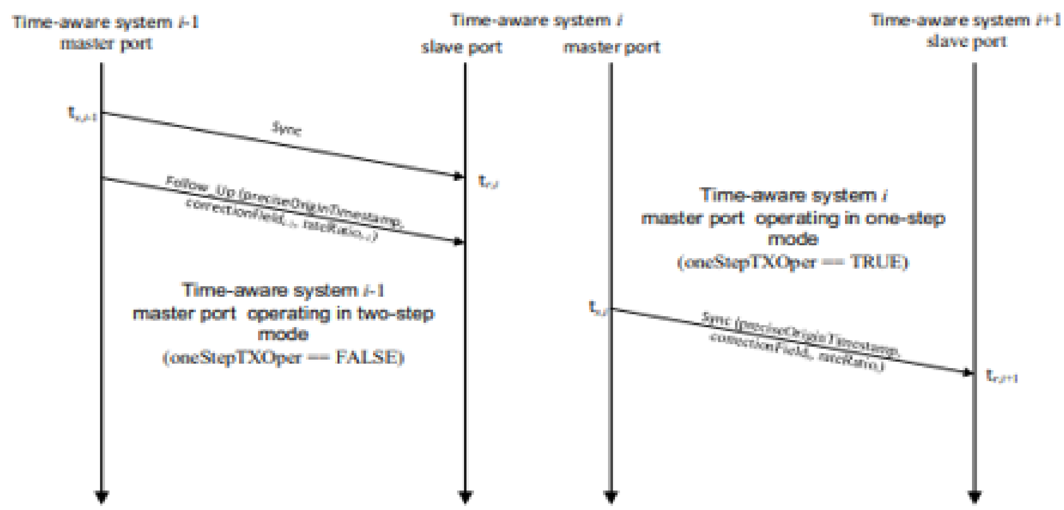


Figure 11-2—Transport of time-synchronization information

(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))

91. The methods practiced by Mitel's use of the accused products include cyclically repeating said frame. For example, the accused products are used by Mitel 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))

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

93. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

94. 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

95. 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."

96. 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.

97. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel MiCollab, Mitel MiCloud Connect, and Mitel (Shoretel) Unified Communications families of products that include advanced presence information capabilities (collectively, "accused products").

ENTERPRISE COLLABORATION SOFTWARE & TOOLS: MICOLLAB

Powering communications for when and where you need it

With MiCollab, your organization has everything it needs to connect, communicate and collaborate across blended environments—driving the exchange of thoughts and improving the speed and quality of decision-making.

(Source : https://www.mitel.com/products/applications/collaboration/micollab)

MiCollab Presence

The MiCollab Client presence feature allows users to monitor other users on the system.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)

MICLOUD CONNECT

Your all-in-one cloud communications, collaboration and contact center service

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure

KEY BENEFITS

WORK FROM ANYWHERE

With Mitel hosted PBX phone systems, you can practically take your system on the go with our smartphone mobile application. With it, you can keep working away from the office without sacrificing your in-office functionality.

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure.

GET A QUOTE

COLLABORATION

Bring teams together with messaging, conferencing, screen sharing, file sharing and more. User can stay connected while on the go with our web, mobile app and desktop applications.

CONTACT CENTER

Choose from our advanced, over-the-top contact center solution, MiCloud Connect CX, or simple, integrated service, MiCloud Connect Contact Center.

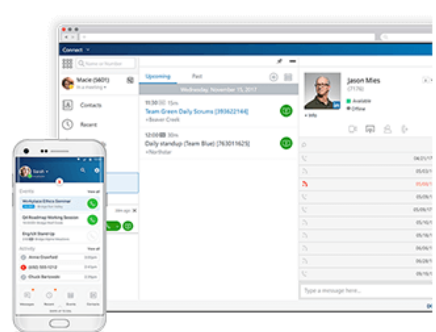
VOICE

Rich PBX features, advanced call controls and softphone, mobile and IP desk phone options so you can talk and meet from anywhere effortlessly.

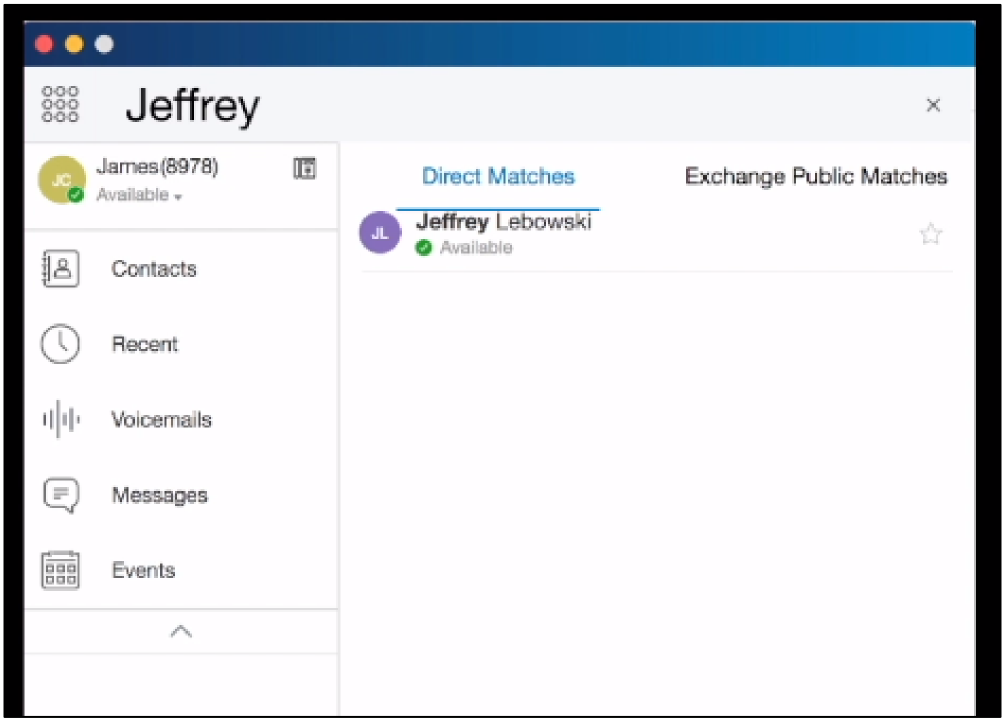
(Source : https://www.mitel.com/en-ca/voip/micloud-connect/features)

EXCEPTIONAL USER EXPERIENCE

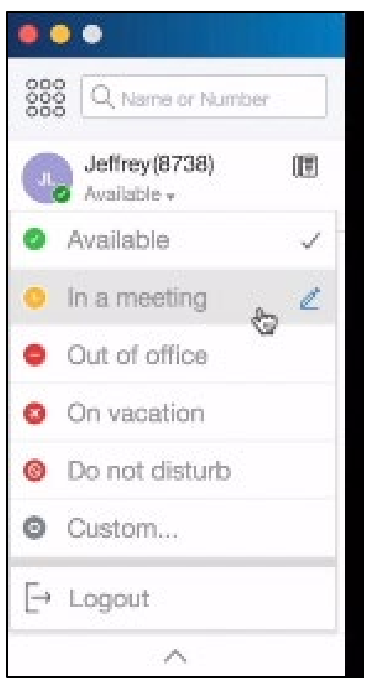
Cloud-based communications and collaboration that make everyday meetings and interactions easier than ever with enterprise-grade VoIP phone services, instant messaging, audio and web conferencing and multi-point video. All with powerful, yet simple, administration for IT teams.



(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)



(Source : http://learn.mitel.com/watch/vKE5mBs5JS2q1MTv15uz1b)



(Source : http://learn.mitel.com/watch/SvkKLJQueZsNLikDQ7tVNc)


ShoreTel
Brilliantly simple™

SOLUTION BRIEF

ShoreTel Unified
Communications Platform

UNIQUE ARCHITECTURE DISTRIBUTES INTELLIGENCE THROUGHOUT THE SYSTEM
FOR SUPERIOR RELIABILITY, FLEXIBILITY AND EASE

(Source : https://microage.com/wp-content/uploads/2016/02/ShoreTel-SOLUTION_UC_Platform_050310.pdf)

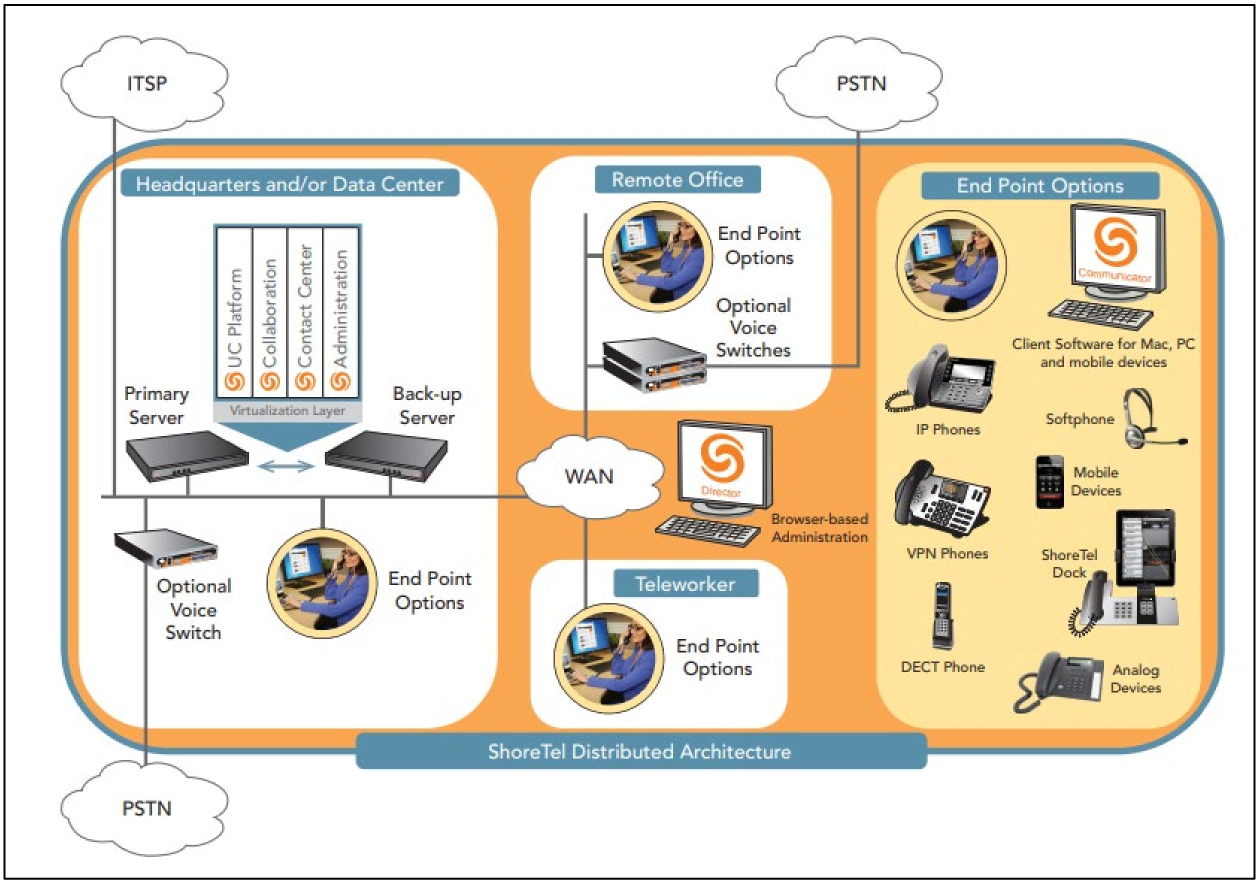
In September 2017, Mitel completed the acquisition of Shoretel.

(Source : https://www.mitel.com/articles/what-happened-shoretel-products)

ShoreTel is the leading provider of brilliantly simple Unified Communications (UC) solutions, including VoIP PBX and business phone systems, enterprise contact centers, collaboration tools and mobile UC solutions.

At the core of ShoreTel's IP business phone solution is our signature Unified Communications (UC) Platform—a unique modular architecture that distributes intelligence across all system to deliver 99.999 percent (five-nines) availability and brilliantly simple system management. Purpose-built for IP, this open, highly resilient UC Platform fits right in with your existing infrastructure, integrates seamlessly with your business applications and processes, and makes unified business communication easy to deploy, manage and scale.

(Source : https://microage.com/wp-content/uploads/2016/02/ShoreTel-SOLUTION_UC_Platform_050310.pdf)



(Source : https://microage.com/wp-content/uploads/2016/02/ShoreTel-SOLUTION_UC_Platform_050310.pdf)

Contact Viewer		
Telephony presence	•	•
Detailed telephony presence (ringing status, connect time)	•	
Instant messaging presence	•	•
Instant messaging presence change alert	•	•
Presence privacy management	•	•
Person-to-person and multi-party IM	•	•
Client-side IM logging	•	•

(Source : http://media.shoretel.com/documents/SPEC_IP_Telephony_US_NP_EN_W_850-1381-01.pdf)

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

99. Mitel 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 a collaboration system for organizations. The accused products provide a communication path to a Private Branch Exchange (PBX) system, a collaboration server, and other integrated applications. The accused products’ service components PbxProxy, Proxy5K are used to monitor the status of desk phones of users. The accused products may also be operated in LAN mode, wherein all the PCs and desk phones of the users are connected to the client server through a LAN, indicating that the client server monitors the status of users’ desk phones via LAN. The presence server component provides the users with the status of the phones of other users. For example, the status that the phone is busy (i.e., active in an ongoing call) is monitored by the accused products.

MiCollab Client Service Components

The MiCollab Client product includes the following MiCollab Client Service components:

- **ADEPM**: Manages Active Directory communication for account synchronization.
- **CSTAProxy**: For MiVoice 5000, MiVoice MX-One, and MiVoice Office 400 deployments only. Manages logging for MiVoice 5000, MiVoice MX-ONE, and MiVoice Office 400 deployments only.
- **DSM**: Analyzes accounts in the PBX database or Active Directory and maintains the account representation in the MiCollab Client Service.
- **EPM**: Manages MiXML-based communication with the MiVoice Business PBX for account synchronization.
- **FEDERATIONGW**: Handles the XMPP federation with third-party systems such as Skype for Business or IBM Lotus Sametime Server.

- **IM**: Handles Instant Messaging between Desktop Clients and provides page mode, conversation mode, and conference mode instant messages.
- **JBoss**: Provides the various administrator features and Web services.
- **PbxProxy**: Maintains MiTAI connections and receives call and feature events from the MiVoice Business PBX. Publishes the events on the MiCollab Client Service internal message bus.
- **Proxy5k**: Maintains OAI connections and receives call and feature events from the MiVoice Office 250 PBX. Publishes the events to the MiCollab Client Service internal message bus.
- **Presence**: Handles subscriptions and notifications for presence, calls, message waiting and so on.
- **RPS**: Includes the server component for the MiCollab Client Service peered connection. This component "listens" on TCP port 36009 for incoming connection requests from the RTC component on a peered MiCollab Client Service.
- **RTC**: Includes the client component for the MiCollab Client Service peered connection. This component connects to the RPS component on a peered MiCollab Client Service.
- **SEE**: Provides the advanced call processing services such as preferential contact call routing.
- **SIPProxy**: Receives SIP messages from the network and routes them to the corresponding MiCollab Client components, such as the SIP Registrar, Presence and IM.
- **SIPRegistrar**: Manages the SIP registrations from the MiCollab Client and notifies other MiCollab Client components whenever registration is added or removed.
- **Watchdog**: Maintains and monitors other MiCollab Client Service components.
- **WSP**: Web Socket Proxy handles the connections from the MiCollab for Mobile for real-time notifications.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)

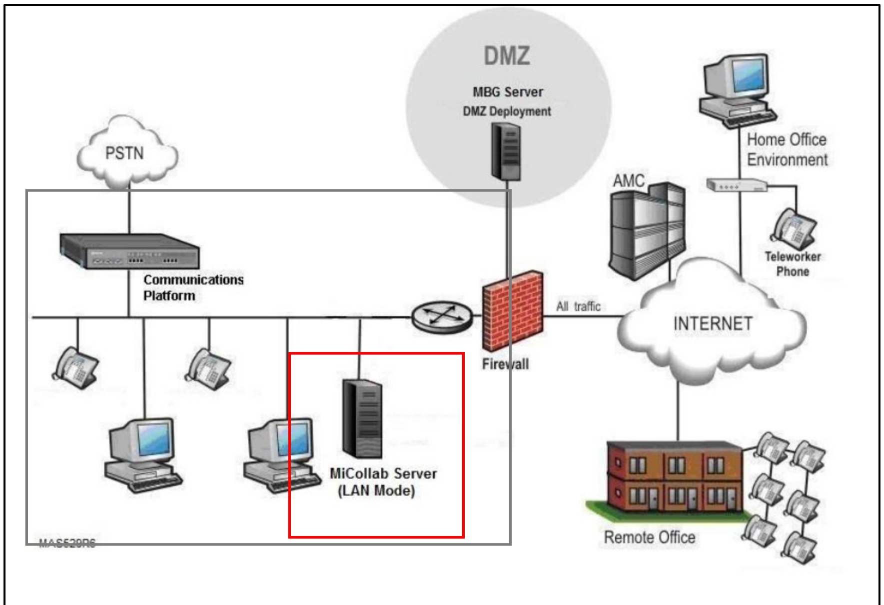


Figure 1: MiCollab in LAN Mode with MBG and Web Proxy in DMZ

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2/MiCollab/MAS_EG_8.1.2.pdf)

MiCollab Client with Web Proxy

A "MiCollab with MiCollab Client in LAN mode (server-only)" configuration requires an MBG server in the DMZ to provide a web proxy. The MBG application on the MiCollab server only provides a single point of provisioning to the MBG server.

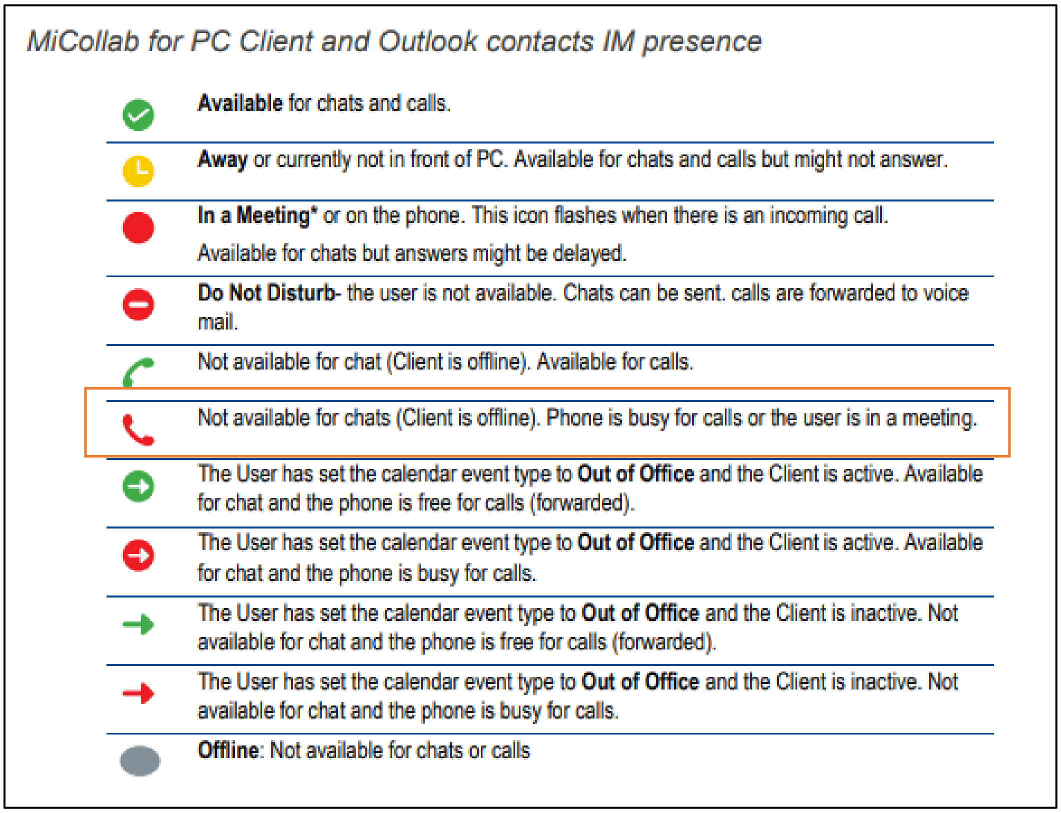
(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2/MiCollab/MAS_EG_8.1.2.pdf)

Presence server

Presence is provided by the Presence Server component on MiCollab Client Service and consists of the following components which provide presence for MiCollab Client users.

- **SIP Proxy**: A SIP-compliant proxy server that routes all the incoming SIP requests to the correct components in MiCollab Client Service.
- **SIP Subscription Manager**: Abstracts the SIP SUBSCRIBE/NOTIFY semantics from the application and implements the application-specific logic.
- **IM Server**: Maintains state information for offline IM messages and conferences. The IM server uses the SIP Subscription Manager to track incoming SIP SUBSCRIBE requests for offline IM and conference states. The SIP Subscription Manager also sends the corresponding SIP NOTIFY requests to subscribers when it receives state changes from the IM Server.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)



MiCollab for PC Client and Outlook contacts IM presence

	Available for chats and calls.
	Away or currently not in front of PC. Available for chats and calls but might not answer.
	In a Meeting* or on the phone. This icon flashes when there is an incoming call. Available for chats but answers might be delayed.
	Do Not Disturb- the user is not available. Chats can be sent. calls are forwarded to voice mail.
	Not available for chat (Client is offline). Available for calls.
	Not available for chats (Client is offline). Phone is busy for calls or the user is in a meeting.
	The User has set the calendar event type to **Out of Office** and the Client is active. Available for chat and the phone is free for calls (forwarded).
	The User has set the calendar event type to **Out of Office** and the Client is active. Available for chat and the phone is busy for calls.
	The User has set the calendar event type to **Out of Office** and the Client is inactive. Not available for chat and the phone is free for calls (forwarded).
	The User has set the calendar event type to **Out of Office** and the Client is inactive. Not available for chat and the phone is busy for calls.
	Offline: Not available for chats or calls

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)

100. The methods practiced by Mitel'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, features of the accused products can be accessed by Android and iOS mobile users using the accused products. This provides mobile users with the presence

feature, wherein the telephony, instant messaging and dynamic status of other users can be transmitted to the mobile user via internet. The accused products provide the mobile user with the status of other users. When a user searches for a contact in the contact list (attempts to call), icons are presented that represent the status of the user. The accused products exchange information from external devices (like teleworker or mobile users) via internet, indicating that the status of the internal phones (monitored via LAN) is sent to users of the accused products via internet. The first network for monitoring the status (i.e., LAN) and the second network used to transfer the status (i.e., internet) are different.

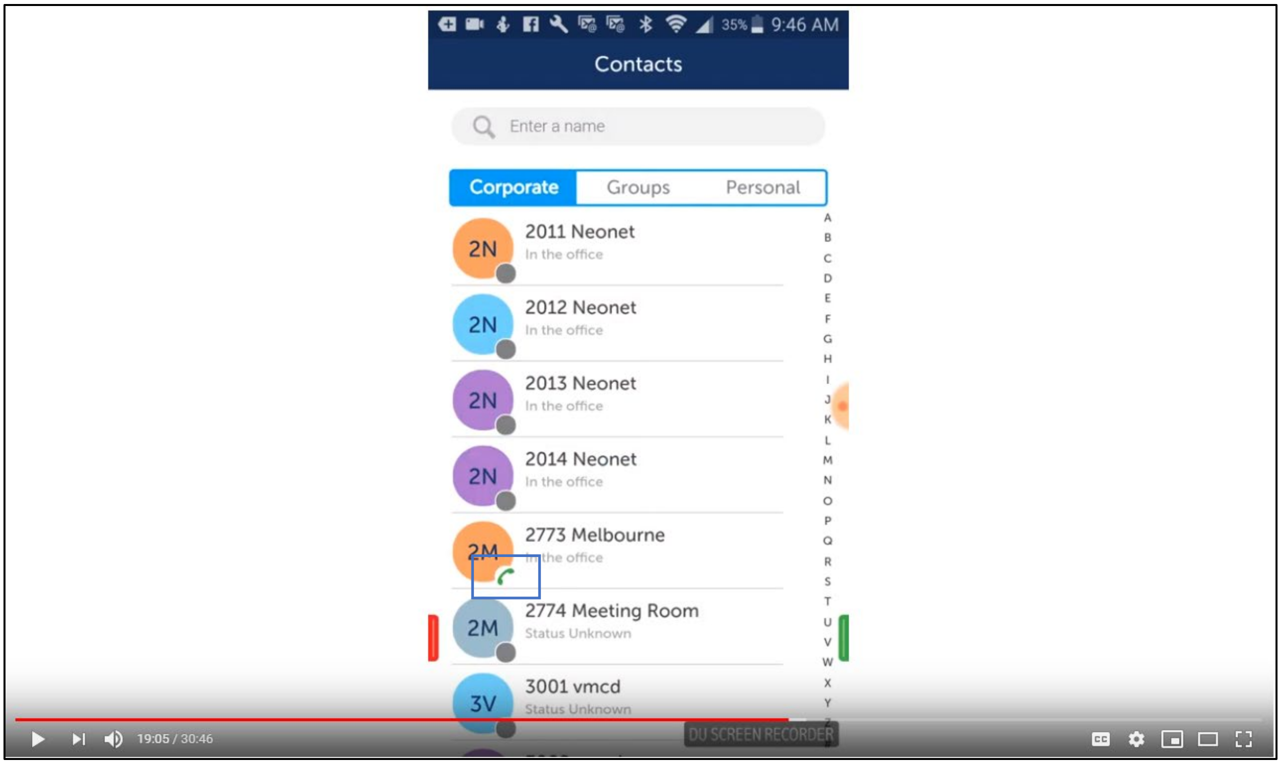
MiCollab for Mobile Client features

Table 3 provides descriptions for the main features accessed from the MiCollab for Mobile Client client.

Table 3: MiCollab for Mobile Client features

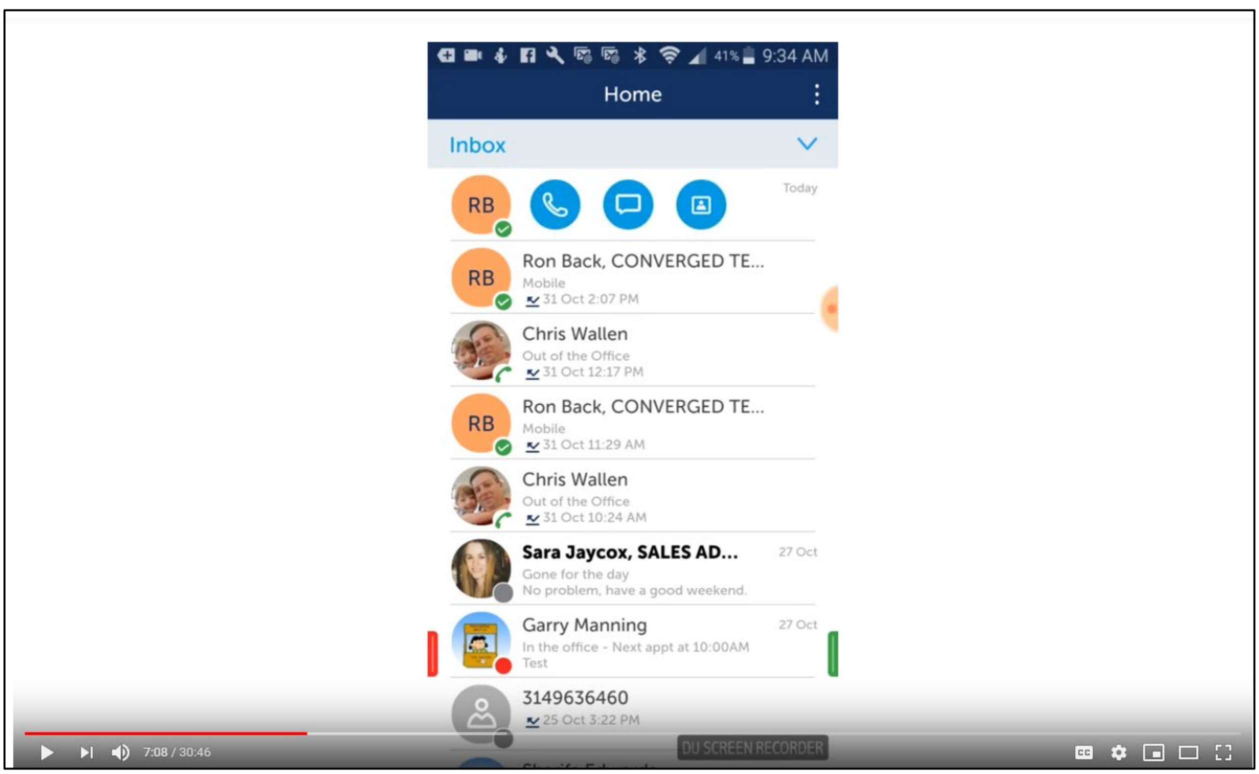
Feature Name	Description	Notes
Ad-hoc Meeting	Ad-hoc Meeting is an instant MiCollab AWV Conference created on all MiCollab Desktop Clients (MiCollab for PC Client, MiCollab MAC Desktop Client, and MiCollab Web Client).	MiCollab for Mobile Client users (Android and iOS) cannot create an ad-hoc meeting. If invited, users will be able to join the meeting on MiCollab for Mobile Client as participants only.
Call History	The user can access call history for missed, received, and dialed calls.	
Call Take	The Call Take feature allows users to take a call active on one device to another device and continue the call without interruption. To use this feature, the user must dial the Feature Access Code from the device on which to continue the call.	MiVoice 5000, MiVoice MX-ONE, and MiVoice Office 400 platforms only.
Chat	Provides multi-party chat functionality for corporate contacts. Chat features include timestamp, file transfer, and chat history.	
Calendar Integration	Integrates MiCollab Client with either your Google, Office 365, or Exchange Server. Regardless of whether or not you are logged into your Calendar or Microsoft Outlook, your MiCollab Client Dynamic Status can access your calendar information directly from the server and update your Dynamic Status appropriately.	
Contacts	Allows the user to access all corporate contact and personal contact information.	
Dynamic Status	Dynamic Status provides customized call routing for MiCollab Client MiCollab Client users. It includes selecting device for outgoing calls and routing incoming calls to preferred device.	
Messages	The user can call voice mail, view, and play received messages.	
MiTeam	MiTeam is Mitel's Cloud-based collaboration tool that provides mobile users with the ability to access features, such as • **Collaborate**: Manage collaboration streams • **Chat**: Hold chat sessions and receive chat notifications • **Pages**: add white-board pages • **To-Do**: Create to-do lists • **File Sharing**: Store and share files, and • **MiTeam Meet:** Perform audio and web sharing within a team.	
Presence	Telephony, Instant Messaging and dynamic status presence.	

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)



(Source: screenshot from video available at

https://www.youtube.com/watch?v=HSoGNAFeAWA)



(Source: screenshot from video available at

https://www.youtube.com/watch?v=HSoGNAFeAWA)

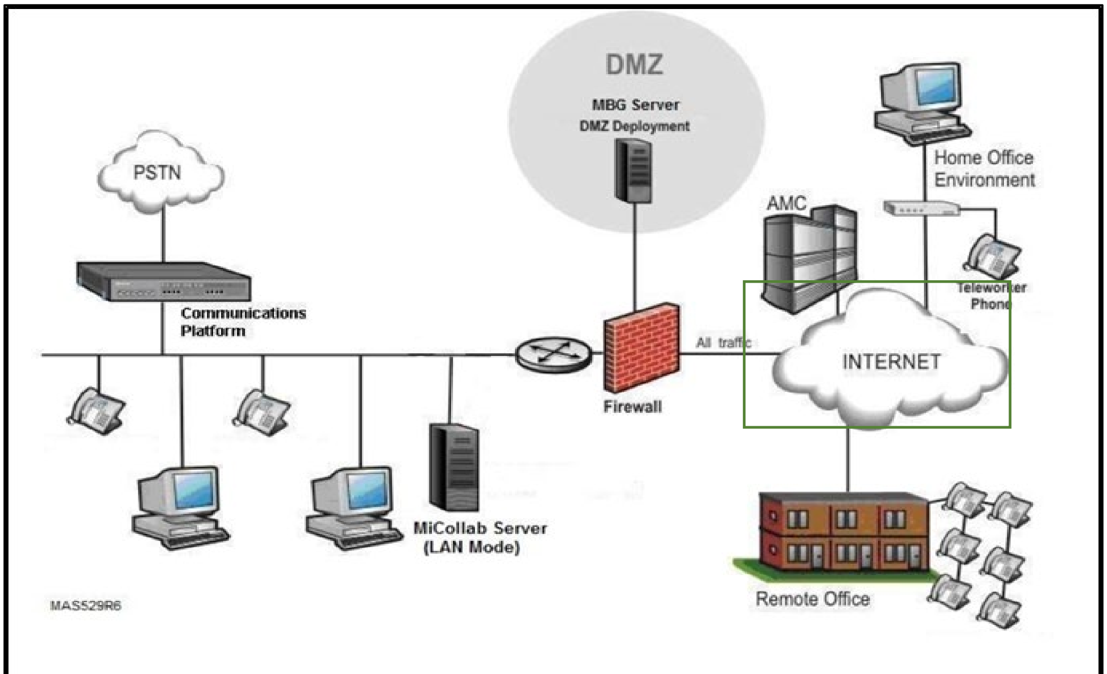


Figure 1: MiCollab in LAN Mode with MBG and Web Proxy in DMZ

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2/MiCollab/MAS_EG_8.1.2.pdf)

Table 2: Firewall Settings for MiCollab with Web Proxy (No Teleworker)

PORT RANGE	DIRECTION	DETAILS
TCP 22 (SSH)	Web Proxy Server → Internet MiCollab Server → Internet	**AMC Communications**. Allow outbound packets (and replies) on TCP port 22 between the Web Proxy and MiCollab Server and the Internet to enable AMC communications (i.e., enable server registration, software and license key downloads, alerts and reporting).
TCP 80 (HTTP)	Web Proxy Server ← Internet	**Web Browser Access**. Allow inbound packets and replies on TCP port 80 between the Web Proxy server and the Internet. Used for remote web browser pages; will be redirected to TCP port 443 (HTTPS).
TCP 443 (HTTPS)	Web Proxy Server ← Internet Web Proxy Server→ LAN	**Web Browser Access**. Allow inbound and outbound packets on TCP port 443 between the Web Proxy server and the Internet for web pages (SSL mode). Allow inbound and outbound packets on TCP port 443 between the Web Proxy server and the LAN for web pages.
TCP 4443 (default – can be configured in web proxy)	Web Proxy Server ← Firewall Web Proxy Server→ LAN	**MiCollab AWV Collaboration Client**. Allow inbound packets on TCP port 443 (default is 443, for single IP allow inbound packets on TCP port 4443) and forward them to configured port (default 4443) on the Web Proxy server as well as return traffic. Allow inbound packets on TCP port 4443 between the Web Proxy server and the LAN. Used for Connection Point traffic related to MiCollab AWV Web Collaboration.
UDP 53 (DNS)	MiCollab Server → Internet Web Proxy Server→ Internet	**Domain Name System.** The server requires DNS to look up the IP address of the Mitel AMC. Alternatively, the server can be configured to forward all DNS requests to another DNS server. See the *MSL Installation and Administration Guide* for details.
TCP 443 (HTTPS)	MiCollab Server → Internet MiCollab Server → Internet	**Mobile Client Deployment.** Used to send deployment tokens and the configuration download URLs to the Mitel

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2/MiCollab/MAS_EG_8.1.2.pdf)

101. Far North Patents only asserts method claims from the '770 Patent.

102. Mitel has had knowledge of the '770 Patent at least as of the date when it was notified of the filing of this action.

103. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

104. 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

105. 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."

106. 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.

107. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel MiCollab, Mitel MiCloud Connect, and Mitel (ShoreTel) Unified Communications families of products that include advanced presence information capabilities (collectively, "accused products"):

ENTERPRISE COLLABORATION SOFTWARE & TOOLS: MICOLLAB

Powering communications for when and where you need it

With MiCollab, your organization has everything it needs to connect, communicate and collaborate across blended environments—driving the exchange of thoughts and improving the speed and quality of decision-making.

(Source : https://www.mitel.com/products/applications/collaboration/micollab)

MiCollab Presence

The MiCollab Client presence feature allows users to monitor other users on the system.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf
)

MICLOUD CONNECT

Your all-in-one cloud communications, collaboration and contact center service

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure

KEY BENEFITS

WORK FROM ANYWHERE

With Mitel hosted PBX phone systems, you can practically take your system on the go with our smartphone mobile application. With it, you can keep working away from the office without sacrificing your in-office functionality.

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure.

GET A QUOTE

COLLABORATION

Bring teams together with messaging, conferencing, screen sharing, file sharing and more. User can stay connected while on the go with our web, mobile app and desktop applications.

CONTACT CENTER

Choose from our advanced, over-the-top contact center solution, MiCloud Connect CX, or simple, integrated service, MiCloud Connect Contact Center.

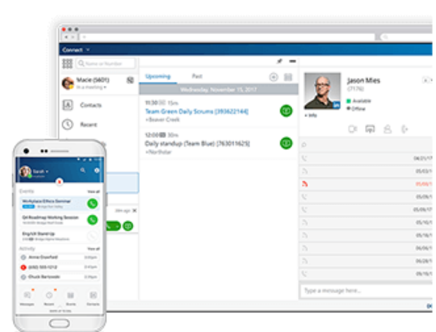
VOICE

Rich PBX features, advanced call controls and softphone, mobile and IP desk phone options so you can talk and meet from anywhere effortlessly.

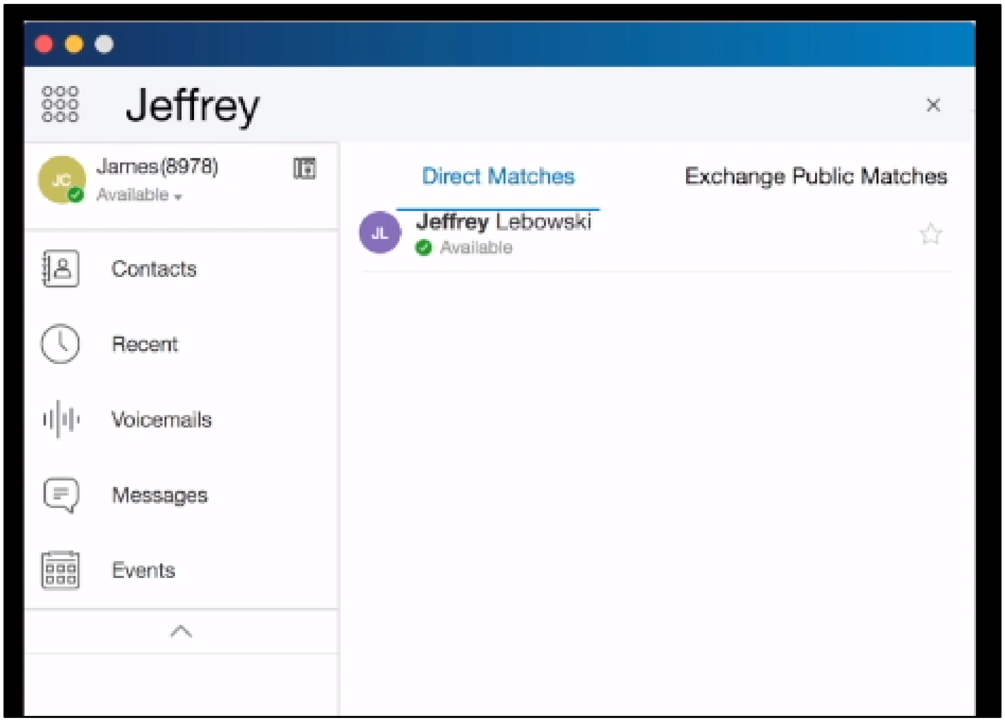
(Source : https://www.mitel.com/en-ca/voip/micloud-connect/features)

EXCEPTIONAL USER EXPERIENCE

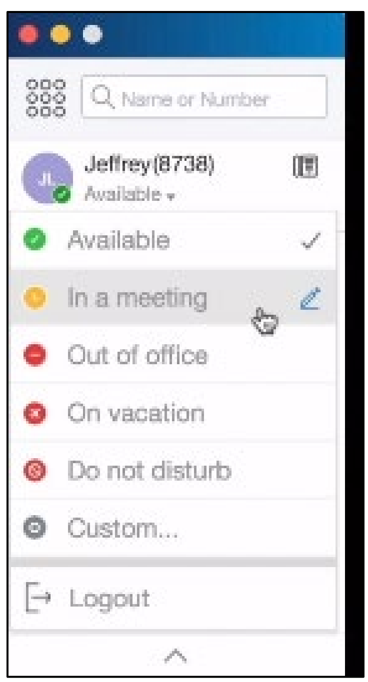
Cloud-based communications and collaboration that make everyday meetings and interactions easier than ever with enterprise-grade VoIP phone services, instant messaging, audio and web conferencing and multi-point video. All with powerful, yet simple, administration for IT teams.



(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)



(Source : http://learn.mitel.com/watch/vKE5mBs5JS2q1MTv15uz1b)



(Source : http://learn.mitel.com/watch/SvkKLJQueZsNLikDQ7tVNc)


ShoreTel
Brilliantly simple™

SOLUTION BRIEF

ShoreTel Unified Communications Platform

UNIQUE ARCHITECTURE DISTRIBUTES INTELLIGENCE THROUGHOUT THE SYSTEM FOR SUPERIOR RELIABILITY, FLEXIBILITY AND EASE

(Source : https://microage.com/wp-content/uploads/2016/02/ShoreTel-SOLUTION_UC_Platform_050310.pdf)

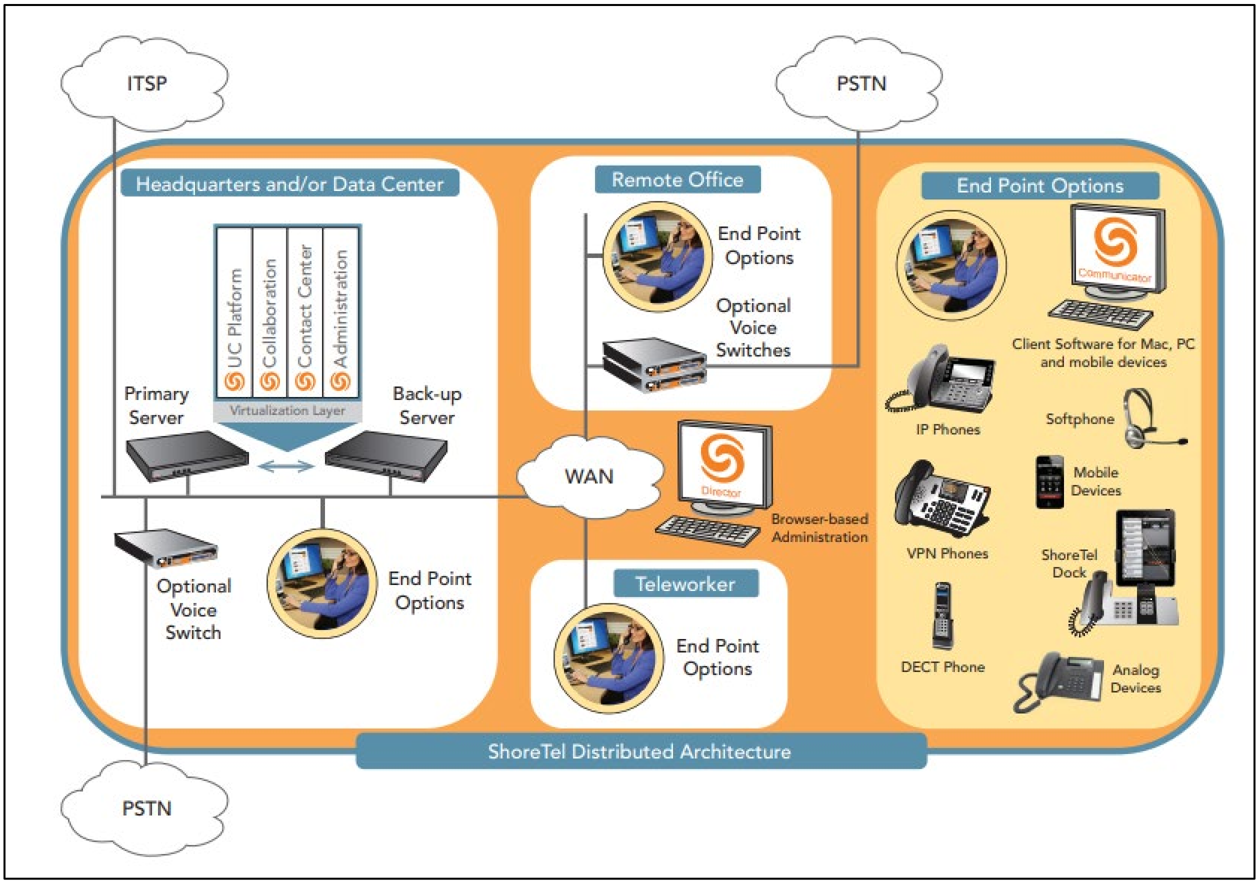
In September 2017, Mitel completed the acquisition of Shoretel.

(Source : https://www.mitel.com/articles/what-happened-shoretel-products)

ShoreTel is the leading provider of brilliantly simple Unified Communications (UC) solutions, including VoIP PBX and business phone systems, enterprise contact centers, collaboration tools and mobile UC solutions.

At the core of ShoreTel's IP business phone solution is our signature Unified Communications (UC) Platform—a unique modular architecture that distributes intelligence across all system to deliver 99.999 percent (five-nines) availability and brilliantly simple system management. Purpose-built for IP, this open, highly resilient UC Platform fits right in with your existing infrastructure, integrates seamlessly with your business applications and processes, and makes unified business communication easy to deploy, manage and scale.

(Source : https://microage.com/wp-content/uploads/2016/02/ShoreTel-SOLUTION_UC_Platform_050310.pdf)



(Source : https://microage.com/wp-content/uploads/2016/02/ShoreTel-SOLUTION_UC_Platform_050310.pdf)

Contact Viewer		
Telephony presence	•	•
Detailed telephony presence (ringing status, connect time)	•	
Instant messaging presence	•	•
Instant messaging presence change alert	•	•
Presence privacy management	•	•
Person-to-person and multi-party IM	•	•
Client-side IM logging	•	•

(Source : http://media.shoretel.com/documents/SPEC_IP_Telephony_US_NP_EN_W_850-1381-01.pdf)

108. By doing so, Mitel has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 5 of the '802 Patent. Mitel's infringement in this regard is ongoing.

109. Mitel 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. For example, the accused products provide collaboration system for organizations. They provide a communication path to a Private Branch Exchange (PBX) system, a collaboration server and other integrated applications. The accused products enable mobile users to use MiCollab Client presence feature. By enabling the GSM line status visibility in an android device using MiCollab Mobile Client app, the presence/availability status of the android device can be seen by other MiCollab Client users. And, the presence server component of the MiCollab client is responsible for presence features including telephony presence status. This indicates that the GSM line status file of the android device is present in the presence server, to enable other users to see the GSM line status of the android device. Further, the MiCollab server (including presence server) exchanges information with mobile users via internet, indicating that the status file is accessed via internet.

MiCollab Presence

The MiCollab Client presence feature allows users to monitor other users on the system.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)

Displaying GSM line status on Android devices

By default, this option is disabled. Enable this option on the MiCollab Client: **Settings** > **Call Settings** to make your GSM line status visible to other MiCollab users.

In the MiCollab Client, the presence indicator on your avatar indicates the line state status. If this option is enabled, your MiCollab line state displays **busy** as soon as your phone engages a GSM call.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.1/MiCollab%20Client/MiCollab_Client_QRG_MOB_8.1.1_en_US.pdf)

Presence server

Presence is provided by the Presence Server component on MiCollab Client Service and consists of the following components which provide presence for MiCollab Client users.

- **SIP Proxy**: A SIP-compliant proxy server that routes all the incoming SIP requests to the correct components in MiCollab Client Service.
- **SIP Subscription Manager**: Abstracts the SIP SUBSCRIBE/NOTIFY semantics from the application and implements the application-specific logic.
- **IM Server**: Maintains state information for offline IM messages and conferences. The IM server uses the SIP Subscription Manager to track incoming SIP SUBSCRIBE requests for offline IM and conference states. The SIP Subscription Manager also sends the corresponding SIP NOTIFY requests to subscribers when it receives state changes from the IM Server.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)

MiCollab for Mobile Client features

Table 3 provides descriptions for the main features accessed from the MiCollab for Mobile Client client.

Table 3: MiCollab for Mobile Client features

Feature Name	Description	Notes
Ad-hoc Meeting	Ad-hoc Meeting is an instant MiCollab AWV Conference created on all MiCollab Desktop Clients (MiCollab for PC Client, MiCollab MAC Desktop Client, and MiCollab Web Client).	MiCollab for Mobile Client users (Android and iOS) cannot create an ad-hoc meeting. If invited, users will be able to join the meeting on MiCollab for Mobile Client as participants only.
Call History	The user can access call history for missed, received, and dialed calls.	
Call Take	The Call Take feature allows users to take a call active on one device to another device and continue the call without interruption. To use this feature, the user must dial the Feature Access Code from the device on which to continue the call.	MiVoice 5000, MiVoice MX-ONE, and MiVoice Office 400 platforms only.
Chat	Provides multi-party chat functionality for corporate contacts. Chat features include timestamp, file transfer, and chat history.	
Calendar Integration	Integrates MiCollab Client with either your Google, Office 365, or Exchange Server. Regardless of whether or not you are logged into your Calendar or Microsoft Outlook, your MiCollab Client Dynamic Status can access your calendar information directly from the server and update your Dynamic Status appropriately.	
Contacts	Allows the user to access all corporate contact and personal contact information.	
Dynamic Status	Dynamic Status provides customized call routing for MiCollab Client MiCollab Client users. It includes selecting device for outgoing calls and routing incoming calls to preferred device.	
Messages	The user can call voice mail, view, and play received messages.	
MiTeam	MiTeam is Mitel's Cloud-based collaboration tool that provides mobile users with the ability to access features, such as • **Collaborate**: Manage collaboration streams • **Chat**: Hold chat sessions and receive chat notifications • **Pages**: add white-board pages • **To-Do**: Create to-do lists • **File Sharing**: Store and share files, and • **MiTeam Meet:** Perform audio and web sharing within a team.	
Presence	Telephony, Instant Messaging and dynamic status presence.	

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.1.pdf)

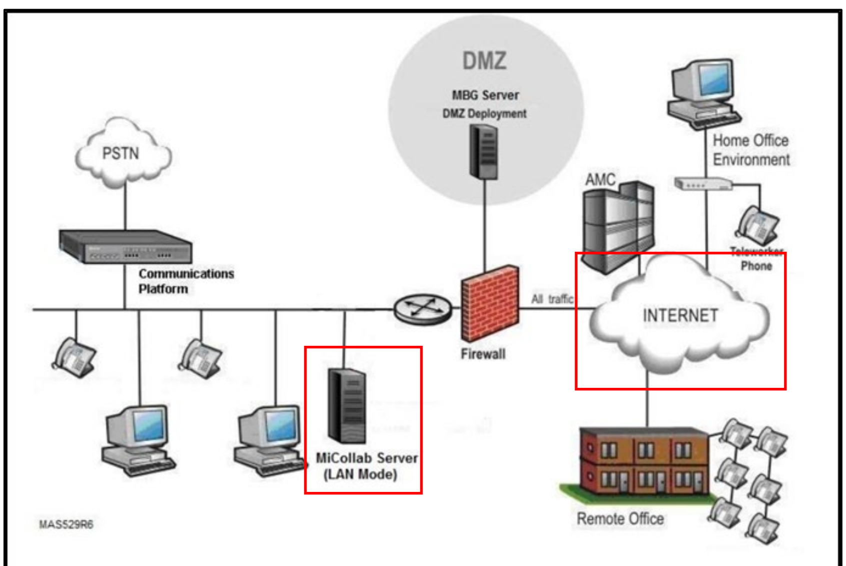


Figure 1: MiCollab in LAN Mode with MBG and Web Proxy in DMZ

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2/MiCollab/MAS_EG_8.1.2.pdf)

Table 2: Firewall Settings for MiCollab with Web Proxy (No Teleworker)

PORT RANGE	DIRECTION	DETAILS
TCP 22 (SSH)	Web Proxy Server → Internet MiCollab Server → Internet	**AMC Communications**. Allow outbound packets (and replies) on TCP port 22 between the Web Proxy and MiCollab Server and the Internet to enable AMC communications (i.e., enable server registration, software and license key downloads, alerts and reporting).
TCP 80 (HTTP)	Web Proxy Server ← Internet	**Web Browser Access**. Allow inbound packets and replies on TCP port 80 between the Web Proxy server and the Internet. Used for remote web browser pages; will be redirected to TCP port 443 (HTTPS).
TCP 443 (HTTPS)	Web Proxy Server ← Internet Web Proxy Server→ LAN	**Web Browser Access**. Allow inbound and outbound packets on TCP port 443 between the Web Proxy server and the Internet for web pages (SSL mode). Allow inbound and outbound packets on TCP port 443 between the Web Proxy server and the LAN for web pages.
TCP 4443 (default – can be configured in web proxy)	Web Proxy Server ← Firewall Web Proxy Server→ LAN	**MiCollab AWV Collaboration Client**. Allow inbound packets on TCP port 443 (default is 443, for single IP allow inbound packets on TCP port 4443) and forward them to configured port (default 4443) on the Web Proxy server as well as return traffic. Allow inbound packets on TCP port 4443 between the Web Proxy server and the LAN. Used for Connection Point traffic related to MiCollab AWV Web Collaboration.
UDP 53 (DNS)	MiCollab Server → Internet Web Proxy Server→ Internet	**Domain Name System.** The server requires DNS to look up the IP address of the Mitel AMC. Alternatively, the server can be configured to forward all DNS requests to another DNS server. See the *MSL Installation and Administration Guide* for details.
TCP 443 (HTTPS)	MiCollab Server → Internet MiCollab Server → Internet	**Mobile Client Deployment.** Used to send deployment tokens and the configuration download URLs to the Mitel

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2/MiCollab/MAS_EG_8.1.2.pdf)

110. The processes practiced by Mitel’s use of the accused products include monitoring the status of the called party's wireless telecommunication device and providing that device status to the device status file. For example, because users of the accused products can see the GSM line status of android devices using the accused products, the GSM line status of the android devices is monitored. The presence server enables the users to see the presence/availability status of other users, indicating that the monitored GSM line status is provided to a presence server, for other users to see the GSM line status of the android phone.

Displaying GSM line status on Android devices

By default, this option is disabled. Enable this option on the MiCollab Client: **Settings** > **Call Settings** to make your GSM line status visible to other MiCollab users.

In the MiCollab Client, the presence indicator on your avatar indicates the line state status. If this option is enabled, your MiCollab line state displays **busy** as soon as your phone engages a GSM call.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.1/MiCollab%20Client/MiCollab_Client_QRG_MOB_8.1.1_en_US.pdf)

MiCollab Presence

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(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.2.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.2.1.pdf)

111. The processes practiced by Mitel'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, when a user of the accused products tries to call an android user (called party), the GSM line status representation of the called party can be seen below the user's avatar, indicating that the status of the called party is sent to the calling party. The status is also presented on a display on a cellular phone, indicating that the status is sent via a pager.

MiCollab for Mobile Client features

Table 3 provides descriptions for the main features accessed from the MiCollab for Mobile Client client.

Table 3: MiCollab for Mobile Client features

Feature Name	Description	Notes
Ad-hoc Meeting	Ad-hoc Meeting is an instant MiCollab AWV Conference created on all MiCollab Desktop Clients (MiCollab for PC Client, MiCollab MAC Desktop Client, and MiCollab Web Client).	MiCollab for Mobile Client users (Android and iOS) cannot create an ad-hoc meeting. If invited, users will be able to join the meeting on MiCollab for Mobile Client as participants only.
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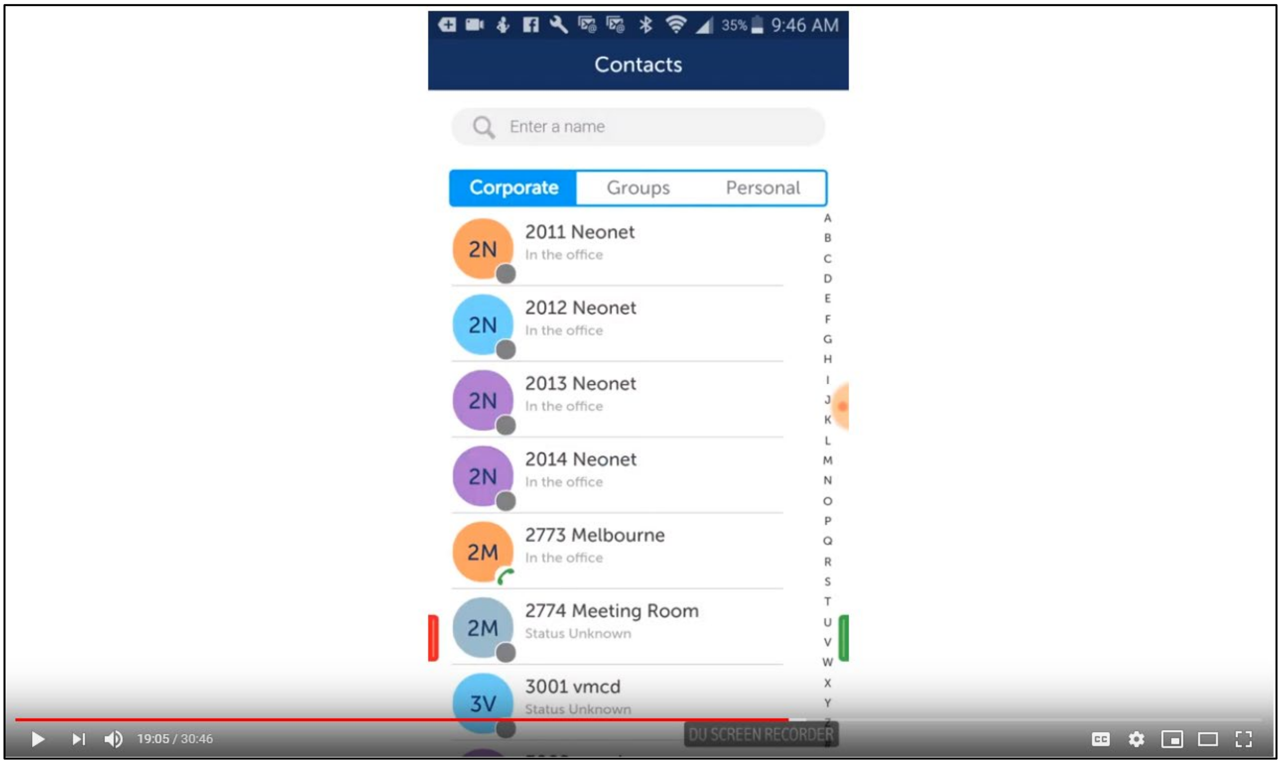
Dynamic Status	Dynamic Status provides customized call routing for MiCollab Client MiCollab Client users. It includes selecting device for outgoing calls and routing incoming calls to preferred device.	
Messages	The user can call voice mail, view, and play received messages.	
MiTeam	MiTeam is Mitel's Cloud-based collaboration tool that provides mobile users with the ability to access features, such as • **Collaborate**: Manage collaboration streams • **Chat**: Hold chat sessions and receive chat notifications • **Pages**: add white-board pages • **To-Do**: Create to-do lists • **File Sharing**: Store and share files, and • **MiTeam Meet:** Perform audio and web sharing within a team.	
Presence	Telephony, Instant Messaging and dynamic status presence.	

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.1/MiCollab%20Client/MiCollab_Client_Admin_Guide_8.1.1.pdf)



(Source: screenshot of video available at https://www.youtube.com/watch?v=HSoGNAFeAWA)

Displaying GSM line status on Android devices

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In the MiCollab Client, the presence indicator on your avatar indicates the line state status. If this option is enabled, your MiCollab line state displays **busy** as soon as your phone engages a GSM call.

(Source: http://edocs.mitel.com/UG/Apps-Solutions/MiCollab%208.1.1/MiCollab%20Client/MiCollab_Client_QRG_MOB_8.1.1_en_US.pdf)

112. Far North Patents only asserts method claims from the '802 Patent.

113. Mitel has had knowledge of the '802 Patent at least as of the date when it was notified of the filing of this action.

114. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

115. 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

116. 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."

117. 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.

118. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel Clearspan, Mitel MiCloud Connect, and Mitel MiCloud Business families of products, that include calling name services for mobile phones (collectively, "accused products"):

MITEL CLEARSPAN

Carrier-Grade Cloud Communications Built to Scale

Mitel Clearspan was built for enterprises, universities, and large institutions looking for a centralized platform with carrier-grade reliability. At its core is a highly scalable call processing architecture capable of serving over 100,000 users without sacrificing ease of management or reliability.

(Source: https://www.mitel.com/products/business-phone-systems/cloud/clearspan)

MICLOUD CONNECT

Your all-in-one cloud communications, collaboration and contact center service

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure

KEY BENEFITS

WORK FROM ANYWHERE

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(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

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COLLABORATION

Bring teams together with messaging, conferencing, screen sharing, file sharing and more. User can stay connected while on the go with our web, mobile app and desktop applications.

CONTACT CENTER

Choose from our advanced, over-the-top contact center solution, MiCloud Connect CX, or simple, integrated service, MiCloud Connect Contact Center.

VOICE

Rich PBX features, advanced call controls and softphone, mobile and IP desk phone options so you can talk and meet from anywhere effortlessly.

(Source : https://www.mitel.com/en-ca/voip/micloud-connect/features)

CONTENT

ShoreTel is now part of Mitel! Powering Connections ® that are Brilliantly Simple ®.

CNAM (Calling Name) is a service in the ShoreTel Sky and Mitel MiCloud Connect phone systems that adds text, such as the name of the caller's company, to the phone number that appears on phones that receive your calls. The text that is added to a phone number can include a maximum of 15 characters. The functionality is similar to Caller ID, which only displays the caller's phone number on phones that receive phone calls. CNAM, also referred to as Caller ID Name, enables companies to display both their organization's name or other text and their phone number (if desired) to customers receiving their calls.

(Source : https://oneview.mitel.com/s/article/CNAM?ui-force-components-controllers-recordGlobalValueProvider.RecordGvp.getRecord=1&r=5)

Overview

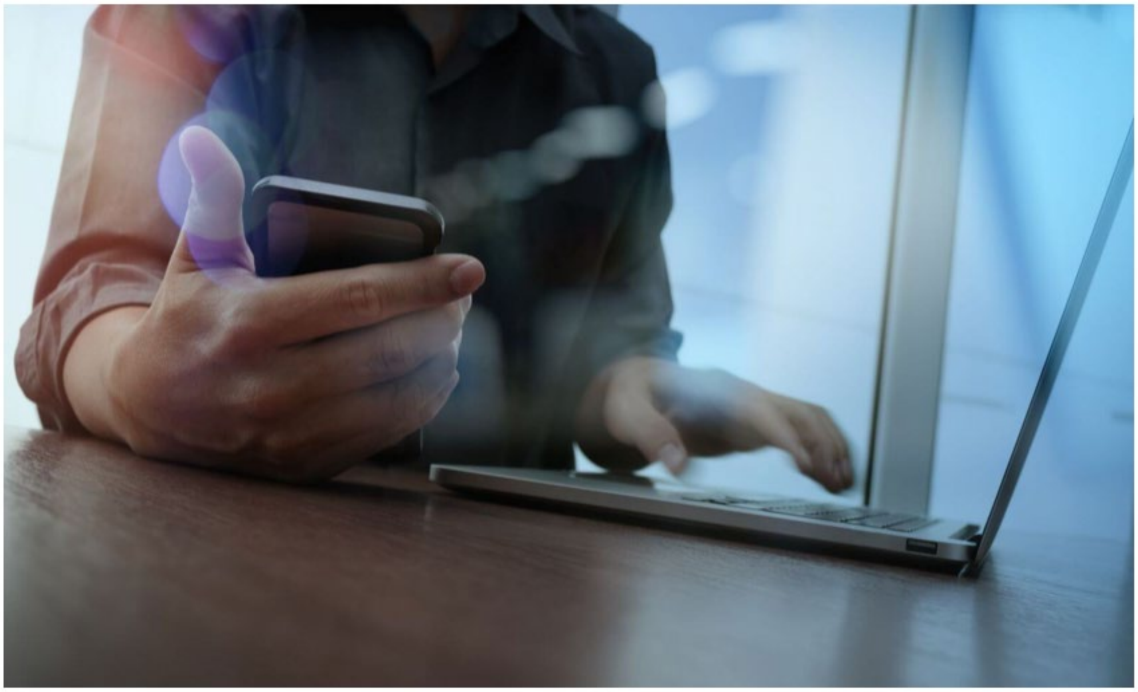
Caller ID, also known as Outbound Caller ID, is a telephone service for residential and small business customers. Within your phone system, your Outbound Caller ID normally displays the main line phone number of your organization or location, but can be changed to display other phone numbers. To manage Outbound Caller ID, see the **Managing Outbound Caller ID** section of this article.

The following are Caller ID-related features of your phone system:

- **Blocked Caller ID** is when phones display "Unknown" or "Anonymous" when a call is received. By default, Caller ID is not blocked for all phone profiles placing outbound calls from your phone system. Authorized Contacts can change this setting in the ShoreTel Sky Portal (see the **Block Caller ID** section of this article). Users can utilize "star" codes to block or unblock Caller ID on a per call basis (see the **Blocking Caller ID Using Your Phone** section of this article).
- **Internal Caller ID** is the name that is displayed on phones when an internal 4-digit extension dial call is received. This name can be changed by updating the **Contact Information** name in the user's **Personal Information** screen. The Internal Caller ID name is also listed in the **Employee Directory** and displayed on several administrator screens in the ShoreTel Sky Portal. For other incoming calls, the name displayed on your phone may be associated with a phone number that has been entered into your **Personal Phonebook**, **Company Phonebook**, or **Employee Directory**. Note that Internal Caller ID is also referred to as the Inbound Caller ID when receiving outside calls (see the **Managing Inbound Caller ID** section of this article).
- **CNAM** (Calling Name) is a service that adds text (a maximum of 15 characters), such as the name of a caller's company, to the phone number that appears on phones that receive calls from that company. The calling party's name is displayed along with, or instead of, the calling number. For more information, see the **CNAM** article.

(Source : https://oneview.mitel.com/s/article/Caller-ID)

MICLOUD BUSINESS



(Source : https://www.mitel.com/en-ca/products/business-phone-systems/cloud/other/micloud-business)

Bring your communications and collaboration platform into the cloud for better mobility, quality, simplicity and reliability—all at a lower cost than a premises-based system. You can add offices, users and features easily to grow and customize your communications for a true competitive advantage.

KEY BENEFITS

BE MORE PRODUCTIVE

Bring all of your communications tools into one cloud and one application for a seamless experience on any device, anywhere in the world. Easily manage voice, email and IM from a single screen to reduce management complexity and boost productivity.

SIMPLE AND SECURE

Enjoy a system that's simple to use and manage: there are no boxes to install, no software to maintain. You get high reliability, exceptional quality and enterprise-class security through our state-of-the-art cloud.

(Source : https://www.mitel.com/en-ca/products/business-phone-systems/cloud/other/micloud-business)

FEATURES

- Local phone numbers
- Unlimited local calling
- Free/Unlimited long distance calling in the U.S. and Canada
- Local number portability
- Mobile twinning
- Hot desking
- Corporate auto attendant
- Automatic transition between day/night routing
- Customized music on hold
- Voicemail with email forwarding
- Hunt/Ring groups
- Audio conferencing
- Interoffice 4-digit dialing
- Localized E911
- Call transfer
- Call forwarding
- Call park
- Call hold
- System speed dial
- User speed dial
- Direct page
- Individual record a call
- Do not disturb
- Call history
- Outbound caller ID (name and number)

(Source : https://www.mitel.com/en-ca/products/business-phone-systems/cloud/other/micloud-business)

MiVOICE 5360 IP PHONE

Ideal for the enterprise executive, the MiVoice 5360 IP phone has a large, high-resolution touch display, superior sound quality and plenty of built-in features.

(Source : http://www.totalcomm.com/wp-content/uploads/2016/12/MitelMiCloudBusiness.pdf)

Viewing Caller ID Information

If you are currently connected to an external caller with Caller ID, you can toggle between the caller's name and number. If the name is unavailable, CANNOT ACCESS FEATURE appears.

To show the outside party's name/number:

Press ⊗ (Special), and then dial **379**.

(Source : http://www.settelecom.nl/bost/pdf/Mitel-5360-manual.pdf)

You can use Call Logging to:

- View recent call activity.
- View caller ID information.
- Return or redial calls.

To use Call Logging:

1. Dial **333** or the **LOGS** menu button.
2. Select one of the following options:
   - Press **1** (MISS) or the **MISSED CALLS** menu button for missed calls.
   - Press **2** (RCV) or the **RECEIVED CALLS** menu button for received calls.
   - Press **3** (DL) or the **DIALED CALLS** menu button for dialed calls.
   - Press **4** (CLR) or the **CLEAR LOGS** menu button to clear all entries.
3. Press ▲ (Up) or ▼ (Down) or the **>>** (Next) or **<<** (Previous) menu buttons to scroll through the entries.

   The display shows the party's name and the extension or outside number (if available) and the date and time.

   If no Caller ID information is available, UNKNOWN CALLER appears.

(Source : http://www.settelecom.nl/bost/pdf/Mitel-5360-manual.pdf)

119. By doing so, Mitel has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 7 of the '517 Patent. Mitel's infringement in this regard is ongoing.

120. Mitel 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. For example, the accused products provide a communication platform for organizations. The Clearspan Anywhere feature enables a user to

register a mobile phone with their desk phone. A call can be made using the registered mobile phone (wireless unit). The Calling Name Retrieval service of Clearspan retrieves and provides the caller name associated with a calling mobile phone (wireless unit) to a called party by querying a database (location register - LR). The database is present in an application server component of Clearspan server. The Calling Name Retrieval service uses a network element like a softswitch to query and retrieve the information associated with the calling mobile phone.

CLEARSPAN® ANYWHERE

Clearspan Anywhere allows you to make and receive calls from any device, at any location, with only one phone number, one dial plan, one voice mailbox, and a unified set of features.

You can call your colleagues from your mobile with their four-digit extension, move calls seamlessly from your desk phone to your mobile when you need to take an important call home with you, and move a call from your mobile to your fixed phone so others can listen in on the speaker phone. This (and more) is all part of Clearspan Anywhere solution.

To ensure that your account is set up for Clearspan Anywhere functionality, contact your office administrator.

MAKE BUSINESS CALLS FROM YOUR MOBILE

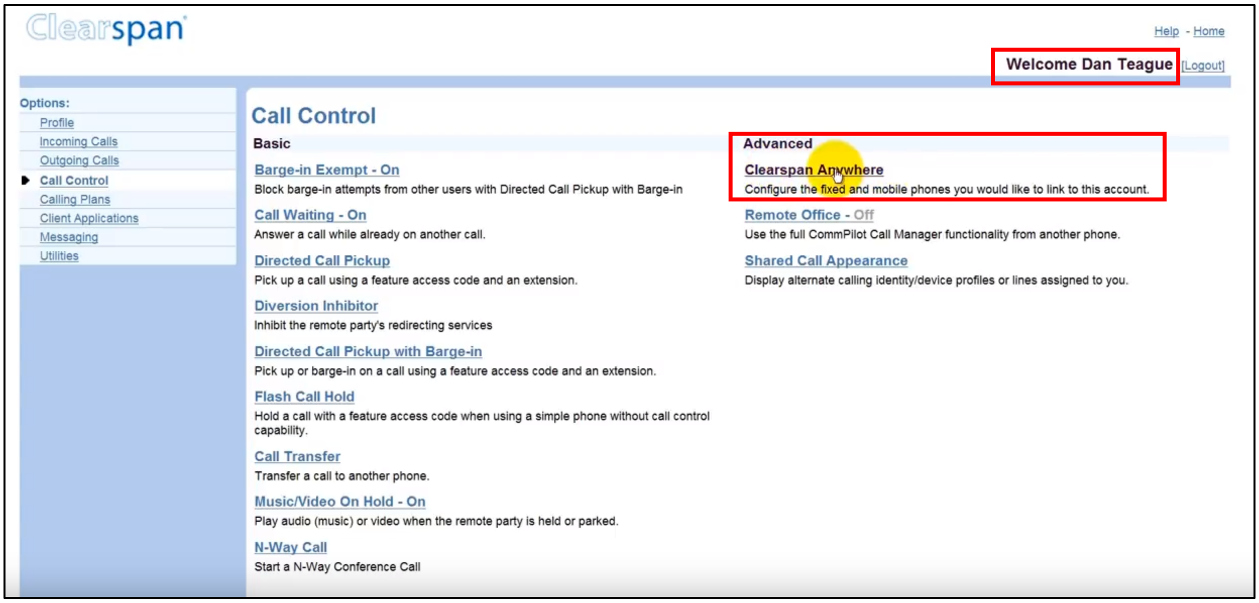
You can make calls from your mobile phone using your Clearspan business number as the calling line ID.

CALL DIRECTLY FROM YOUR MOBILE

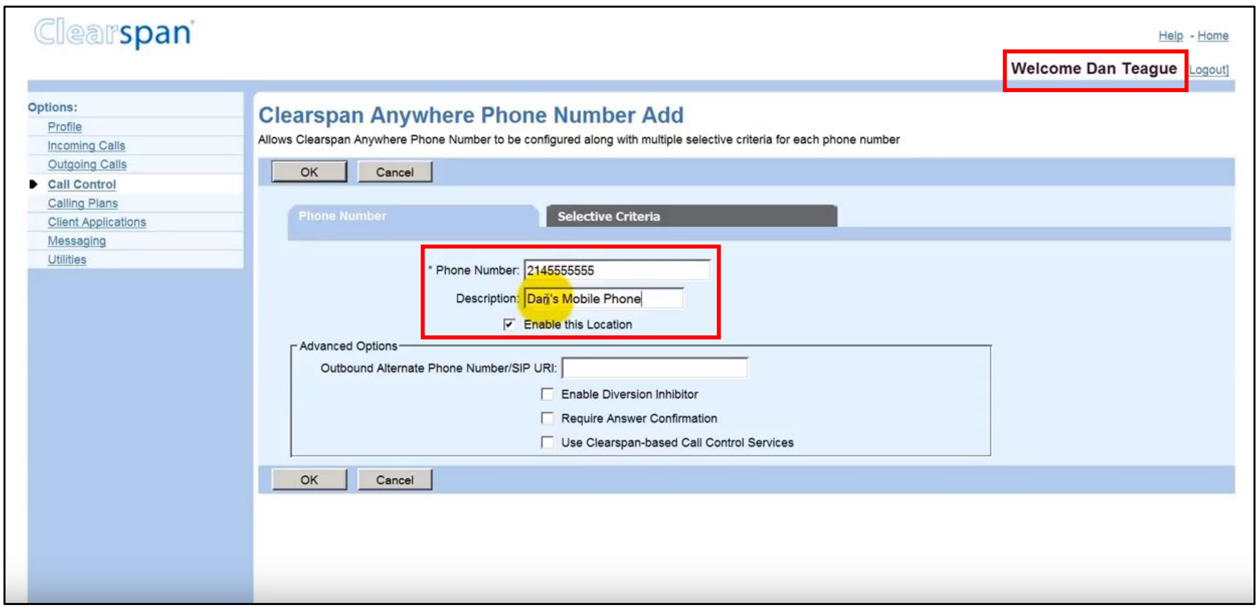
1. From your mobile, dial your Clearspan Anywhere Portal Number.
2. Wait for the Two-Stage dial tone.
3. Once you hear the tone, dial the destination number or business extension. The called party sees your Clearspan business number (not mobile number) as the calling line ID.

(Source:

http://edocs.mitel.com/UG/EN/Clearspan/285906%20CS%20Clearspan%20Anywhere%20QRG%20R22.0.pdf)



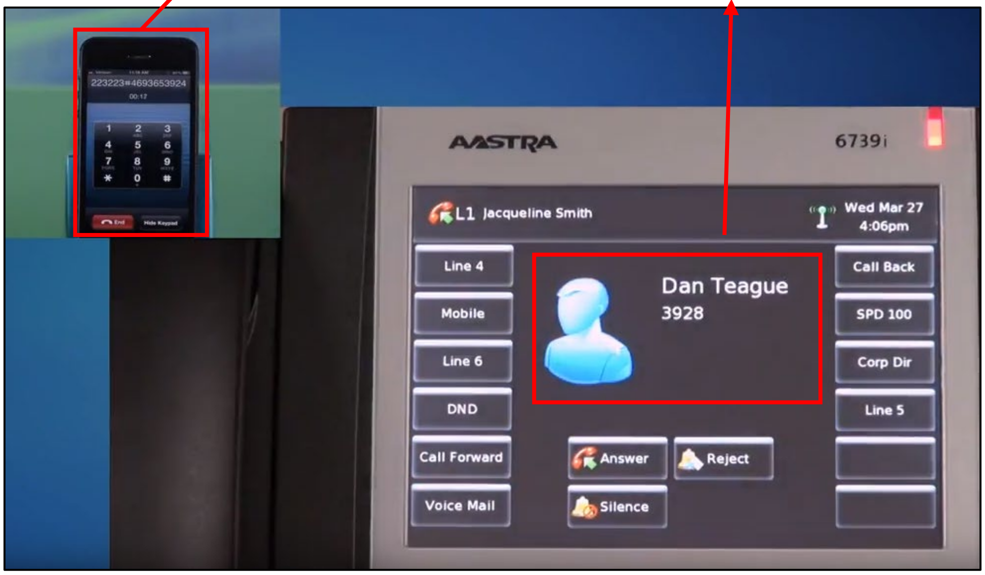
(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

Caller's mobile phone (wireless unit in a wireless network)

Calling party's name on the called party's desk phone (terminating unit of a wireline network)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

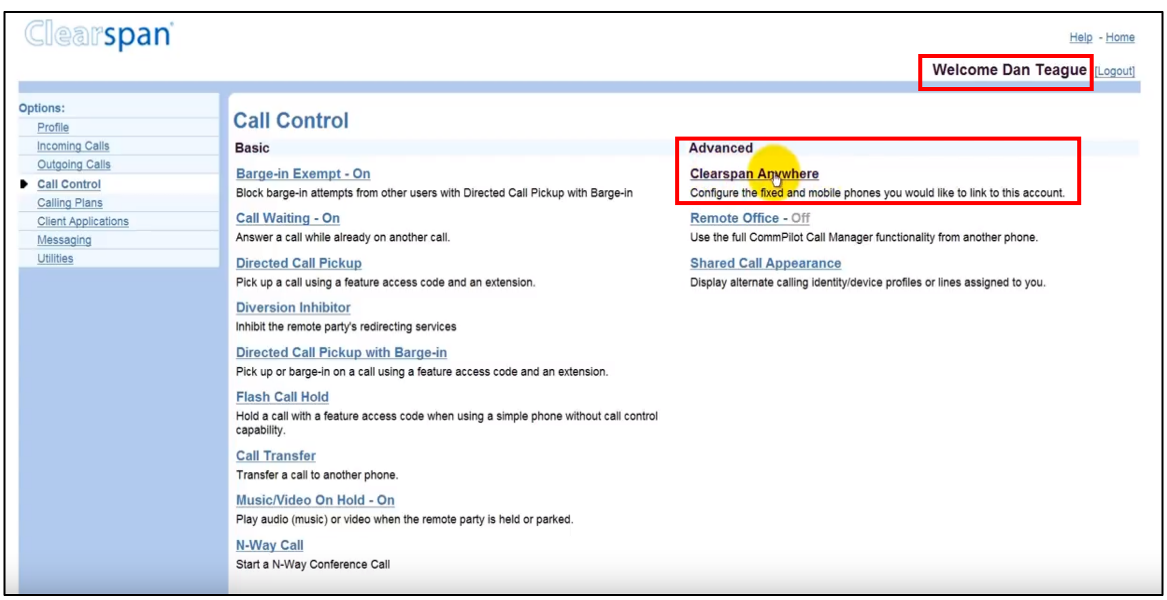
- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

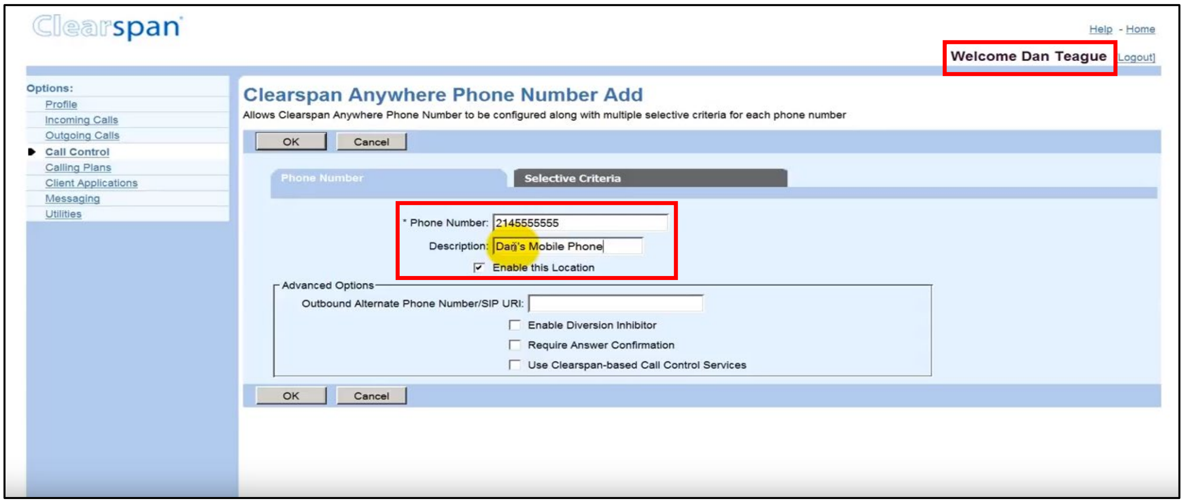
> A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

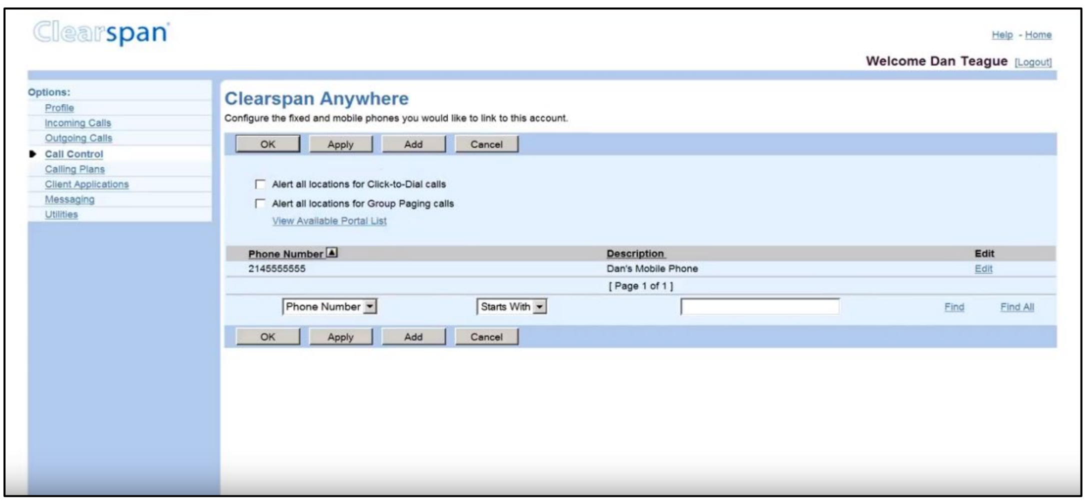
121. The methods practiced by Mitel's use of the accused products include causing the LR to include a plurality of entries with each entry including an identifier and corresponding respectively to a wireless unit, each identifier of an entry being associated with information corresponding respectively to the wireless unit. For example, the Mitel "Clearspan Anywhere" feature enables a user to register a mobile phone with their desk phone. The user provides a mobile phone number that needs to be registered with desk phone by logging into the Clearspan portal. The database (LR) present in the application server of Clearspan server maintains the information of all the users. The database includes multiple entries in which each entry includes a registered mobile phone number (identifier corresponding to the wireless unit). And the registered mobile number is associated with the user's name (identifier being associated with information corresponding to the wireless unit).



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)



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(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

Application Server (AS)

The Application Server (AS) is shown in Figure 4 (previous page). The AS is a service delivery platform responsible for the execution and management of enhanced personal and group services. This server maintains the user database and features assigned to those users. The AS functions also include management of network traffic, handling of signaling interfaces, and logical execution and management of services. The AS comprises a database, the ServiceOS™ abstraction layer, and protocol stacks.

There are multiple layers of service configuration for services delivered from the AS. Secure web access is provided by the Clearspan XSP/Web Server, described in the XSP Server section, which enables management, administration, provisioning, and configuration. The associated web portals can be customized for different user groups, based on the services of those groups.

The Clearspan database maintains user and group profiles, as well as service and subscription data. Updates and access are performed in real time. The ServiceOS manages the sessions, which are the network connections associated with a user.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

Inventory Management

Clearspan maintains a list of all devices that are provisioned in the network. This is integrated into the same database that manages all users, lines, and services in the network. This means Clearspan can easily track relationships between devices, the ports that are free, the ports that are in use, and the corresponding users who are associated with each port on the device. This type of information is invaluable when tracking the state of devices in the network and troubleshooting problems on the access network. Clearspan also provides basic inventory management reporting tools that can be used either by themselves or integrated with a broader inventory management system.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

122. The methods practiced by Mitel'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, using the accused products, the database (LR) present in the application server of the Clearspan server, in response to a received query that includes a caller number (identifier), looks up for the caller's name associated with the caller number.

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

DESCRIPTION

This service provides a SIP subscription-based method or a Simple Object Access Protocol/ Extensible Markup Language (SOAP/XML) query method of retrieving calling name information from an external database on a per-call basis; this function is analogous to the GR-1188 Transactional Capabilities Application Part (TCAP) terminating query.

The basic service works as follows:

- If the name information is already present in the incoming call setup message, then the external database is not accessed.
- If the name information is not available, Clearspan sends a request to the external database.

The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)

123. The methods practiced by Mitel'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. For example, using the accused products, the database (LR) present in the application server of the Clearspan server retrieves the caller name associated with the caller number. The caller name (information) corresponds to a "Clearspan Anywhere" user who has registered a mobile phone (wireless unit) with a desktop phone.

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

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(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

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- If the name information is not available, Clearspan sends a request to the external database.

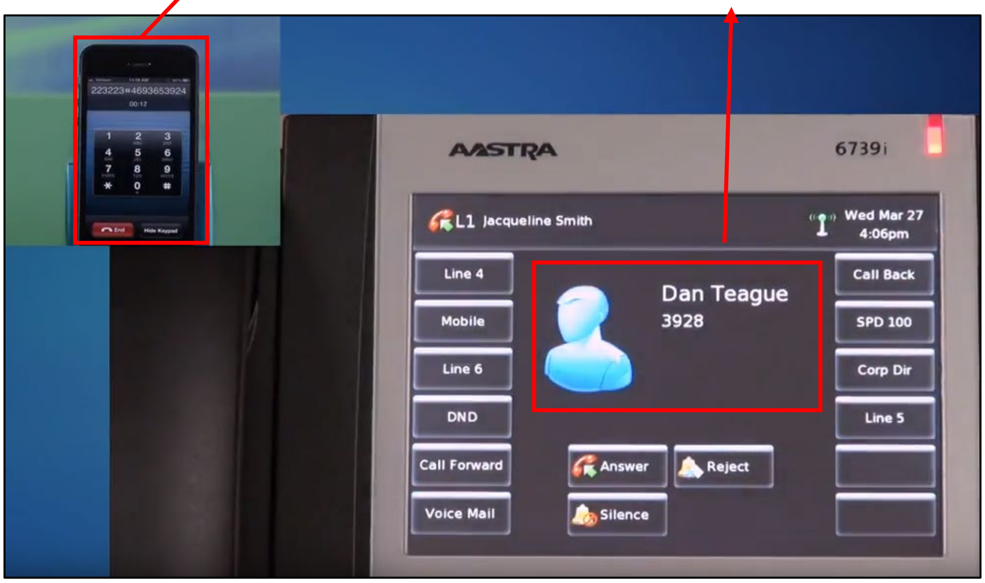
The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)

Caller's mobile phone (wireless unit in a wireless network)

Calling party's name on the called party's desk phone (terminating unit of a wireline network)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

124. The methods practiced by Mitel's use of the accused products include causing the LR to provide the information in a response to the query. For example, using the accused products, the database (LR) present in the application server of the Clearspan server provides the caller name information associated with the caller number. Further, the caller name (for example: Dan Teague) information is presented to the called party.

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

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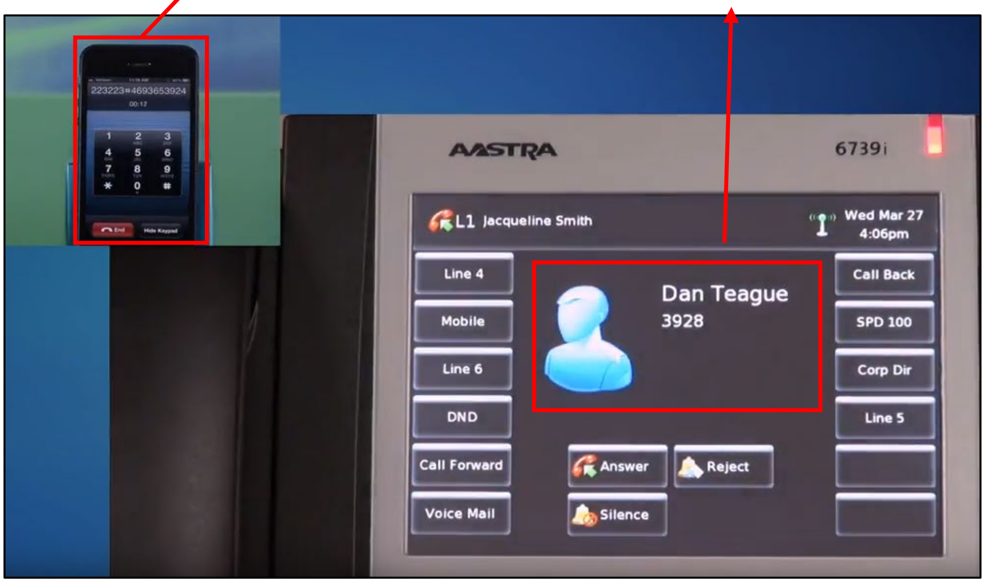
The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)

Caller's mobile phone (wireless unit in a wireless network)

Calling party's name on the called party's desk phone (terminating unit of a wireline network)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

125. Far North Patents only asserts method claims from the '517 Patent.

126. Mitel has had knowledge of the '517 Patent at least as of the date when it was notified of the filing of this action.

127. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

128. 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

129. 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."

130. 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.

131. Mitel made, had made, used, imported, provided, supplied, distributed, sold, and/or offered for sale products and/or systems including, for example, its Mitel Clearspan, Mitel

MiCloud Connect, and Mitel MiCloud Business families of products, that include calling name services for mobile phones (collectively, "accused products"):

MITEL CLEARSPAN

Carrier-Grade Cloud Communications Built to Scale

Mitel Clearspan was built for enterprises, universities, and large institutions looking for a centralized platform with carrier-grade reliability. At its core is a highly scalable call processing architecture capable of serving over 100,000 users without sacrificing ease of management or reliability.

(Source: https://www.mitel.com/products/business-phone-systems/cloud/clearspan)

MICLOUD CONNECT

Your all-in-one cloud communications, collaboration and contact center service

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure

KEY BENEFITS

WORK FROM ANYWHERE

With Mitel hosted PBX phone systems, you can practically take your system on the go with our smartphone mobile application. With it, you can keep working away from the office without sacrificing your in-office functionality.

(Source : https://www.mitel.com/products/business-phone-systems/cloud/micloud-connect)

MiCloud Connect

MiCloud Connect is a complete cloud business communications service that delivers seamless voice, collaboration and contact center solutions from a single provider. By combining an intuitive user experience and flexible service plans with Google Cloud's proven reliability, MiCloud Connect makes every aspect of cloud communications and collaboration simple and secure.

GET A QUOTE

COLLABORATION

Bring teams together with messaging, conferencing, screen sharing, file sharing and more. User can stay connected while on the go with our web, mobile app and desktop applications.

CONTACT CENTER

Choose from our advanced, over-the-top contact center solution, MiCloud Connect CX, or simple, integrated service, MiCloud Connect Contact Center.

VOICE

Rich PBX features, advanced call controls and softphone, mobile and IP desk phone options so you can talk and meet from anywhere effortlessly.

(Source : https://www.mitel.com/en-ca/voip/micloud-connect/features)

CONTENT

ShoreTel is now part of Mitel! Powering Connections ® that are Brilliantly Simple ®.

CNAM (Calling Name) is a service in the ShoreTel Sky and Mitel MiCloud Connect phone systems that adds text, such as the name of the caller's company, to the phone number that appears on phones that receive your calls. The text that is added to a phone number can include a maximum of 15 characters. The functionality is similar to Caller ID, which only displays the caller's phone number on phones that receive phone calls. CNAM, also referred to as Caller ID Name, enables companies to display both their organization's name or other text and their phone number (if desired) to customers receiving their calls.

(Source : https://oneview.mitel.com/s/article/CNAM?ui-force-components-controllers-recordGlobalValueProvider.RecordGvp.getRecord=1&r=5)

Overview

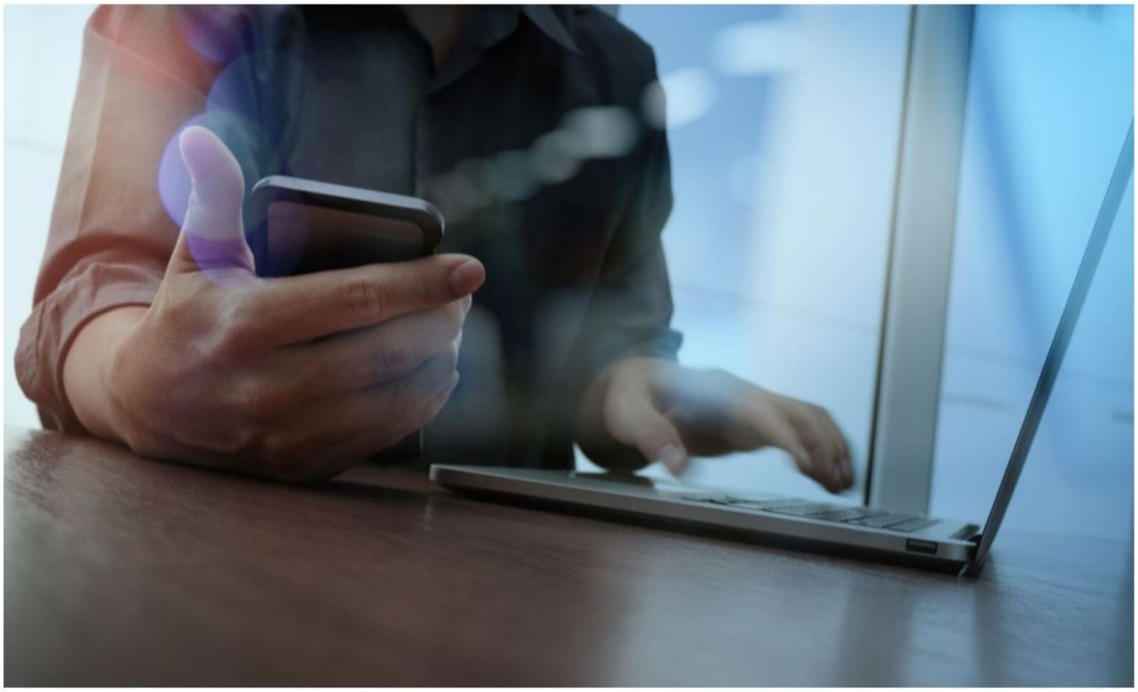
Caller ID, also known as Outbound Caller ID, is a telephone service for residential and small business customers. Within your phone system, your Outbound Caller ID normally displays the main line phone number of your organization or location, but can be changed to display other phone numbers. To manage Outbound Caller ID, see the **Managing Outbound Caller ID** section of this article.

The following are Caller ID-related features of your phone system:

- **Blocked Caller ID** is when phones display "Unknown" or "Anonymous" when a call is received. By default, Caller ID is not blocked for all phone profiles placing outbound calls from your phone system. Authorized Contacts can change this setting in the ShoreTel Sky Portal (see the **Block Caller ID** section of this article). Users can utilize "star" codes to block or unblock Caller ID on a per call basis (see the **Blocking Caller ID Using Your Phone** section of this article).
- **Internal Caller ID** is the name that is displayed on phones when an internal 4-digit extension dial call is received. This name can be changed by updating the **Contact Information** name in the user's **Personal Information** screen. The Internal Caller ID name is also listed in the **Employee Directory** and displayed on several administrator screens in the ShoreTel Sky Portal. For other incoming calls, the name displayed on your phone may be associated with a phone number that has been entered into your **Personal Phonebook**, **Company Phonebook**, or **Employee Directory**. Note that Internal Caller ID is also referred to as the Inbound Caller ID when receiving outside calls (see the **Managing Inbound Caller ID** section of this article).
- **CNAM** (Calling Name) is a service that adds text (a maximum of 15 characters), such as the name of a caller's company, to the phone number that appears on phones that receive calls from that company. The calling party's name is displayed along with, or instead of, the calling number. For more information, see the **CNAM** article.

(Source : https://oneview.mitel.com/s/article/Caller-ID)

MICLOUD BUSINESS



(Source : https://www.mitel.com/en-ca/products/business-phone-systems/cloud/other/micloud-business)

Bring your communications and collaboration platform into the cloud for better mobility, quality, simplicity and reliability—all at a lower cost than a premises-based system. You can add offices, users and features easily to grow and customize your communications for a true competitive advantage.

KEY BENEFITS

BE MORE PRODUCTIVE

Bring all of your communications tools into one cloud and one application for a seamless experience on any device, anywhere in the world. Easily manage voice, email and IM from a single screen to reduce management complexity and boost productivity.

SIMPLE AND SECURE

Enjoy a system that's simple to use and manage: there are no boxes to install, no software to maintain. You get high reliability, exceptional quality and enterprise-class security through our state-of-the-art cloud.

(Source : https://www.mitel.com/en-ca/products/business-phone-systems/cloud/other/micloud-business)

FEATURES

- Local phone numbers
- Unlimited local calling
- Free/Unlimited long distance calling in the U.S. and Canada
- Local number portability
- Mobile twinning
- Hot desking
- Corporate auto attendant
- Automatic transition between day/night routing
- Customized music on hold
- Voicemail with email forwarding
- Hunt/Ring groups
- Audio conferencing
- Interoffice 4-digit dialing
- Localized E911
- Call transfer
- Call forwarding
- Call park
- Call hold
- System speed dial
- User speed dial
- Direct page
- Individual record a call
- Do not disturb
- Call history
- Outbound caller ID (name and number)

(Source : https://www.mitel.com/en-ca/products/business-phone-systems/cloud/other/micloud-business)

MiVOICE 5360 IP PHONE

Ideal for the enterprise executive, the MiVoice 5360 IP phone has a large, high-resolution touch display, superior sound quality and plenty of built-in features.

(Source : http://www.totalcomm.com/wp-content/uploads/2016/12/MitelMiCloudBusiness.pdf)

Viewing Caller ID Information

If you are currently connected to an external caller with Caller ID, you can toggle between the caller's name and number. If the name is unavailable, CANNOT ACCESS FEATURE appears.

To show the outside party's name/number:

Press ⊗ (Special), and then dial **379**.

(Source : http://www.settelecom.nl/bost/pdf/Mitel-5360-manual.pdf)

You can use Call Logging to:

- View recent call activity.
- View caller ID information.
- Return or redial calls.

To use Call Logging:

1. Dial **333** or the **LOGS** menu button.
2. Select one of the following options:
   - Press **1** (MISS) or the **MISSED CALLS** menu button for missed calls.
   - Press **2** (RCV) or the **RECEIVED CALLS** menu button for received calls.
   - Press **3** (DL) or the **DIALED CALLS** menu button for dialed calls.
   - Press **4** (CLR) or the **CLEAR LOGS** menu button to clear all entries.
3. Press ⌃ (Up) or ⌄ (Down) or the **>>** (Next) or **<<** (Previous) menu buttons to scroll through the entries.

   The display shows the party's name and the extension or outside number (if available) and the date and time.

   If no Caller ID information is available, UNKNOWN CALLER appears.

(Source : http://www.settelecom.nl/bost/pdf/Mitel-5360-manual.pdf)

132. By doing so, Mitel has directly infringed (literally and/or under the doctrine of equivalents) at least Claim 5 of the '588 Patent. Mitel's infringement in this regard is ongoing.

133. Mitel 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. For example, the accused products provide a communication platform for organizations. The Clearspan Anywhere feature enables a user to

register a mobile phone with their desk phone. A call can be made using the registered mobile phone (wireless unit). The Calling Name Retrieval service of Clearspan retrieves and provides the caller name associated with a calling mobile phone (wireless unit) to a called party by querying a database (location register - LR). The database is present in an application server component of the Clearspan server. The Calling Name Retrieval service uses a network element like a softswitch to query and retrieve the information associated with the calling mobile phone.

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Clearspan Anywhere allows you to make and receive calls from any device, at any location, with only one phone number, one dial plan, one voice mailbox, and a unified set of features.

You can call your colleagues from your mobile with their four-digit extension, move calls seamlessly from your desk phone to your mobile when you need to take an important call home with you, and move a call from your mobile to your fixed phone so others can listen in on the speaker phone. This (and more) is all part of Clearspan Anywhere solution.

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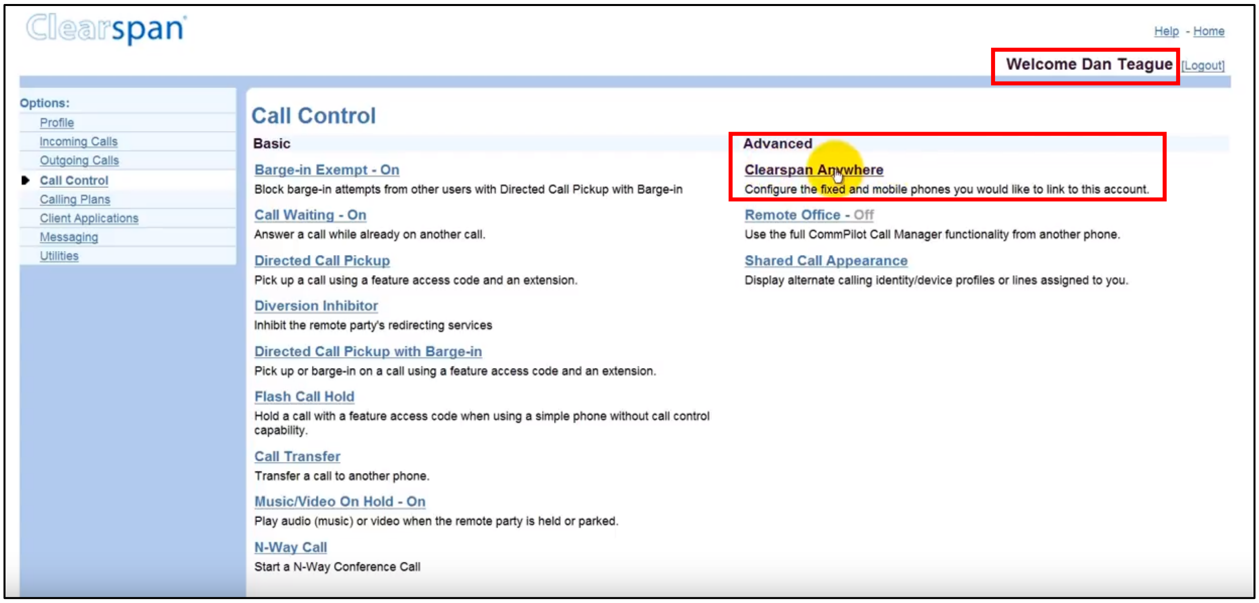
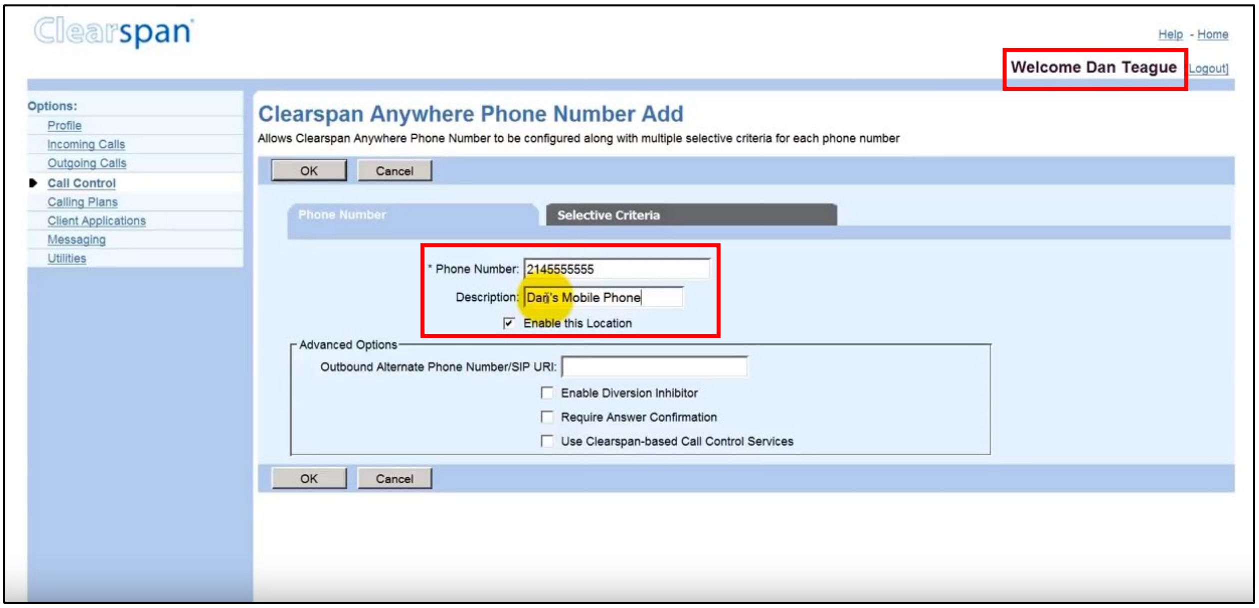
1. From your mobile, dial your Clearspan Anywhere Portal Number.
2. Wait for the Two-Stage dial tone.
3. Once you hear the tone, dial the destination number or business extension. The called party sees your Clearspan business number (not mobile number) as the calling line ID.

(Source:

http://edocs.mitel.com/UG/EN/Clearspan/285906%20CS%20Clearspan%20Anywhere%20QRG%20R22.0.pdf)



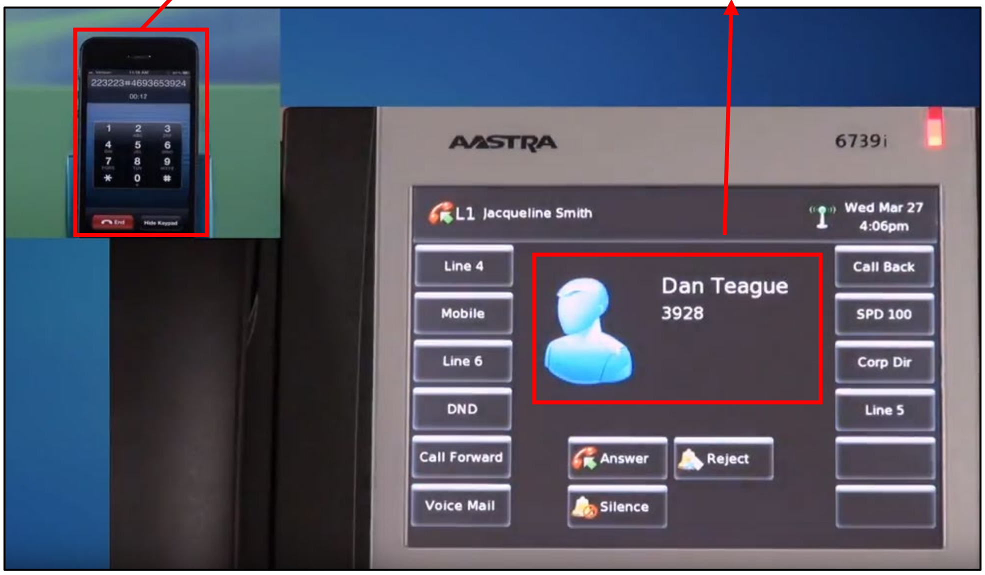
(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

Caller's mobile phone (wireless unit in a wireless network)

Calling party's name on the called party's desk phone (terminating unit of a wireline network)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

> A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

134. The methods practiced by Mitel's use of the accused products include 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 Clearspan Anywhere feature of the accused products enables a user to register a mobile phone with their desk phone. The user provides a mobile phone number that needs to be registered with a desk phone by logging into the Clearspan portal. By maintaining the information of all users, the database (LR) present in the application server of the Clearspan server includes multiple entries in which each entry includes a registered mobile phone number (Mobile Identity Number - MIN corresponding to the wireless unit). The registered mobile number is associated with the user's name (MIN being associated with information corresponding to the wireless unit). The user who is calling has an option to activate or deactivate (presentation allowance) or subscribe for Calling Line ID Delivery Blocking service that decides the presentation of caller name associated with a calling mobile phone (wireless unit) to a called party's device. All the subscriptions of a user are stored in the database of the application server.

CLEARSPAN® ANYWHERE

Clearspan Anywhere allows you to make and receive calls from any device, at any location, with only one phone number, one dial plan, one voice mailbox, and a unified set of features.

You can call your colleagues from your mobile with their four-digit extension, move calls seamlessly from your desk phone to your mobile when you need to take an important call home with you, and move a call from your mobile to your fixed phone so others can listen in on the speaker phone. This (and more) is all part of Clearspan Anywhere solution.

To ensure that your account is set up for Clearspan Anywhere functionality, contact your office administrator.

MAKE BUSINESS CALLS FROM YOUR MOBILE

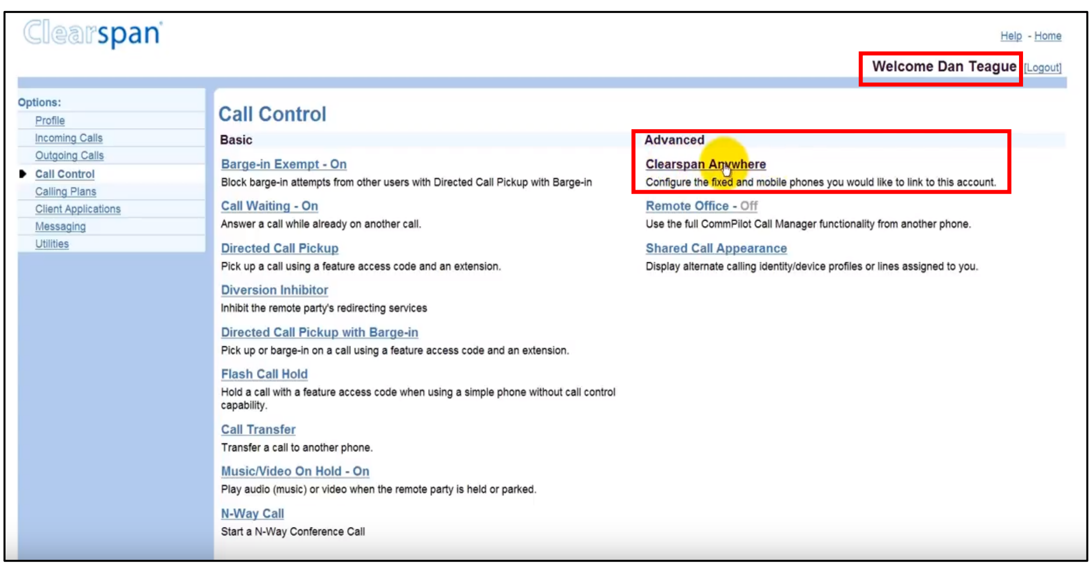
You can make calls from your mobile phone using your Clearspan business number as the calling line ID.

CALL DIRECTLY FROM YOUR MOBILE

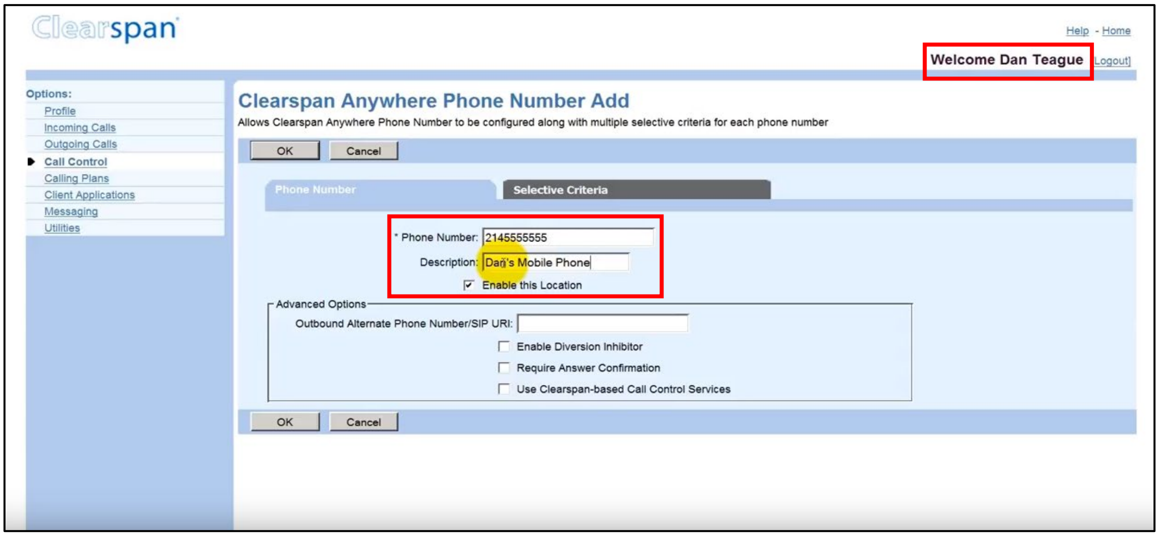
1. From your mobile, dial your Clearspan Anywhere Portal Number.
2. Wait for the Two-Stage dial tone.
3. Once you hear the tone, dial the destination number or business extension. The called party sees your Clearspan business number (not mobile number) as the calling line ID.

(Source:

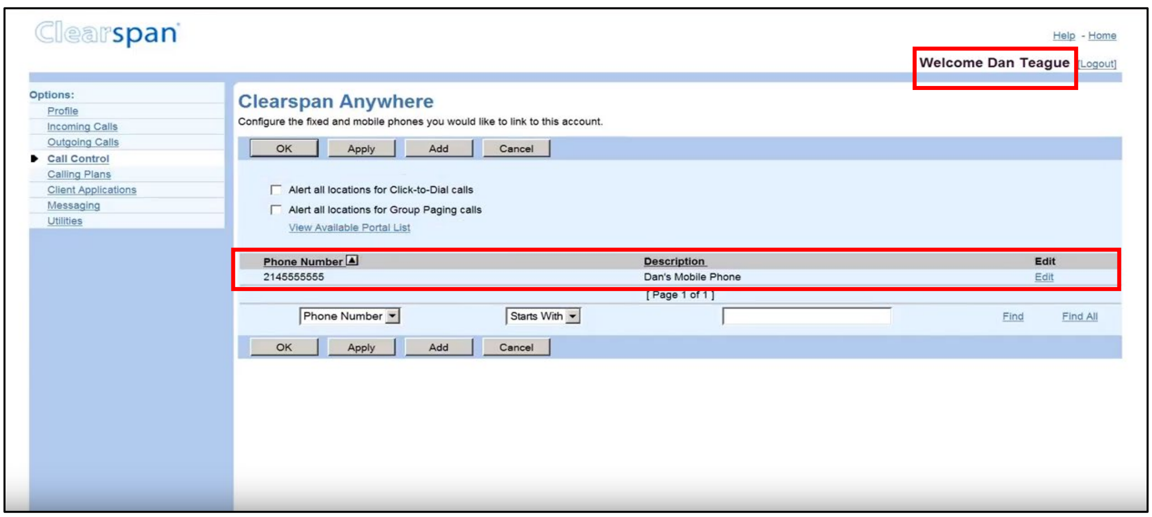
http://edocs.mitel.com/UG/EN/Clearspan/285906%20CS%20Clearspan%20Anywhere%20QRG%20R22.0.pdf)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

CALLING LINE ID DELIVERY BLOCKING

This service enables a user to block delivery of their identity to the called party.

DESCRIPTION

Calling Line ID Delivery Blocking blocks the delivery of a user's identity (both name and number) to a called party.

Calls made by the user to parties outside of the group or enterprise have the presentation of their identity (name and number) blocked.

CONFIGURATION

Users can activate or deactivate the Calling Line ID Delivery Blocking service via the Personal web portal, or by dialing a configurable feature access code from their phone.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/282407.pdf)

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

Application Server (AS)

The Application Server (AS) is shown in Figure 4 (previous page). The AS is a service delivery platform responsible for the execution and management of enhanced personal and group services. This server maintains the user database and features assigned to those users. The AS functions also include management of network traffic, handling of signaling interfaces, and logical execution and management of services. The AS comprises a database, the ServiceOS™ abstraction layer, and protocol stacks.

There are multiple layers of service configuration for services delivered from the AS. Secure web access is provided by the Clearspan XSP/Web Server, described in the XSP Server section, which enables management, administration, provisioning, and configuration. The associated web portals can be customized for different user groups, based on the services of those groups.

The Clearspan database maintains user and group profiles, as well as service and subscription data. Updates and access are performed in real time. The ServiceOS manages the sessions, which are the network connections associated with a user.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

> **Inventory Management**
>
> Clearspan maintains a list of all devices that are provisioned in the network. This is integrated into the same database that manages all users, lines, and services in the network. This means Clearspan can easily track relationships between devices, the ports that are free, the ports that are in use, and the corresponding users who are associated with each port on the device. This type of information is invaluable when tracking the state of devices in the network and troubleshooting problems on the access network. Clearspan also provides basic inventory management reporting tools that can be used either by themselves or integrated with a broader inventory management system.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

135. The methods practiced by Mitel'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. For example, the database (LR) present in the application server of Clearspan server, in response to a received query that includes a caller number (MIN), looks up the caller's name associated with the caller number (MIN).

> **Clearspan Server Components**
>
> Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:
>
> - Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

> A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source: http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

DESCRIPTION

This service provides a SIP subscription-based method or a Simple Object Access Protocol/ Extensible Markup Language (SOAP/XML) query method of retrieving calling name information from an external database on a per-call basis; this function is analogous to the GR-1188 Transactional Capabilities Application Part (TCAP) terminating query.

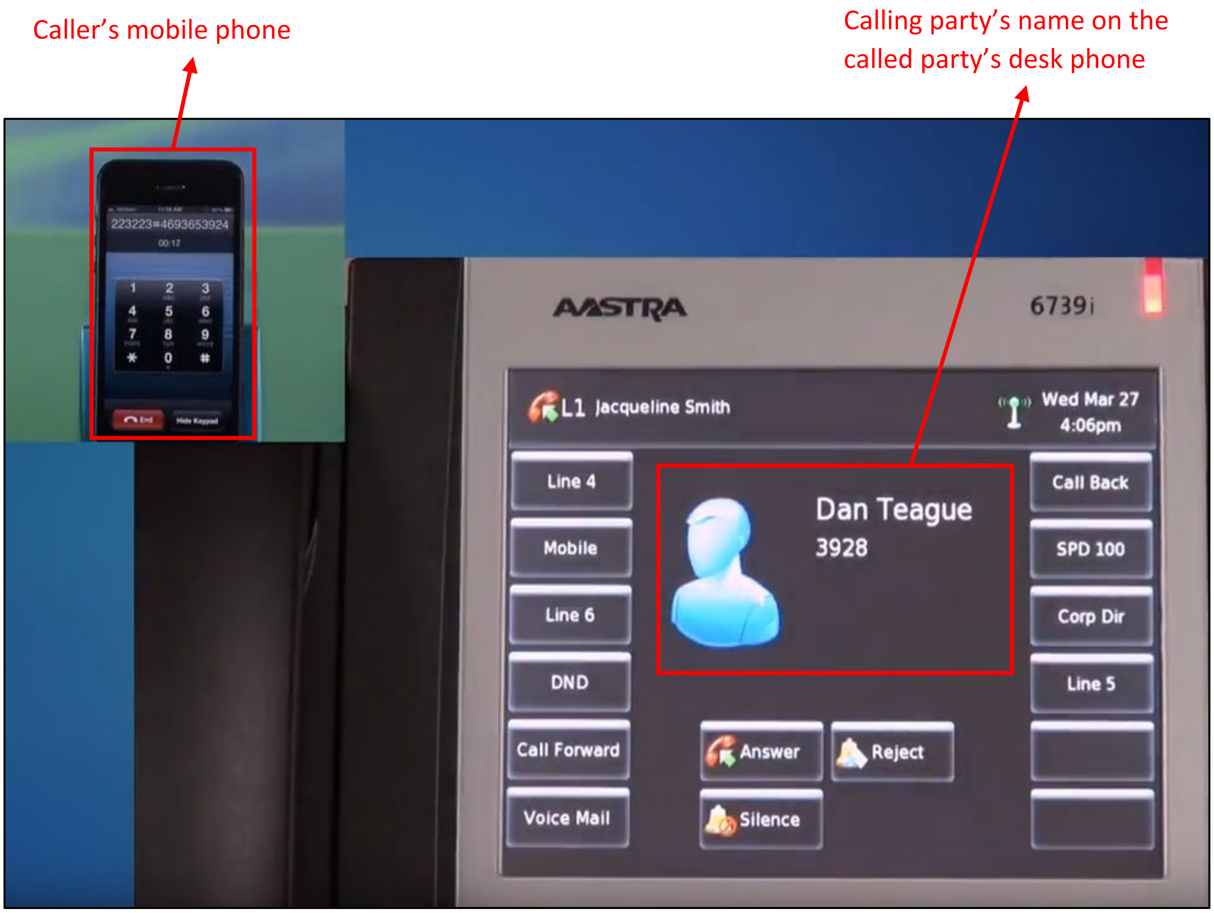
The basic service works as follows:

- If the name information is already present in the incoming call setup message, then the external database is not accessed.
- If the name information is not available, Clearspan sends a request to the external database.

The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

136. The methods practiced by Mitel's use of the accused products include checking that the information associated with the MIN and/or the MDN of the entry comprises the presentation allowance. For example, the Calling Line ID Delivery Blocking service blocks the presentation of calling ID (calling number and calling name) to the called party. The calling user needs to subscribe, for such service to avoid calling ID display on the called party device. All the subscriptions of the users are maintained in the database of the application server. The called party can retrieve the calling name from the application server's database if the calling user has disabled the Calling Line ID Delivery Blocking. The calling name retrieval method of Clearspan

checks for the Calling Line ID Delivery Blocking feature subscription of calling user to retrieve calling name from the database.

Application Server (AS)

The Application Server (AS) is shown in Figure 4 (previous page). The AS is a service delivery platform responsible for the execution and management of enhanced personal and group services. This server maintains the user database and features assigned to those users. The AS functions also include management of network traffic, handling of signaling interfaces, and logical execution and management of services. The AS comprises a database, the ServiceOS™ abstraction layer, and protocol stacks.

There are multiple layers of service configuration for services delivered from the AS. Secure web access is provided by the Clearspan XSP/Web Server, described in the XSP Server section, which enables management, administration, provisioning, and configuration. The associated web portals can be customized for different user groups, based on the services of those groups.

The Clearspan database maintains user and group profiles, as well as service and subscription data. Updates and access are performed in real time. The ServiceOS manages the sessions, which are the network connections associated with a user.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

CALLING LINE ID DELIVERY BLOCKING

This service enables a user to block delivery of their identity to the called party.

DESCRIPTION

Calling Line ID Delivery Blocking blocks the delivery of a user's identity (both name and number) to a called party.

Calls made by the user to parties outside of the group or enterprise have the presentation of their identity (name and number) blocked.

CONFIGURATION

Users can activate or deactivate the Calling Line ID Delivery Blocking service via the Personal web portal, or by dialing a configurable feature access code from their phone.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/282407.pdf)

137. The methods practiced by Mitel'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. For example, the database (LR) present in the application server of Clearspan server retrieves the caller name associated with the caller number. The caller name (information) corresponds to a 'Clearspan Anywhere' user who has registered mobile phone (wireless unit) with desktop phone.

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

DESCRIPTION

This service provides a SIP subscription-based method or a Simple Object Access Protocol/ Extensible Markup Language (SOAP/XML) query method of retrieving calling name information from an external database on a per-call basis; this function is analogous to the GR-1188 Transactional Capabilities Application Part (TCAP) terminating query.

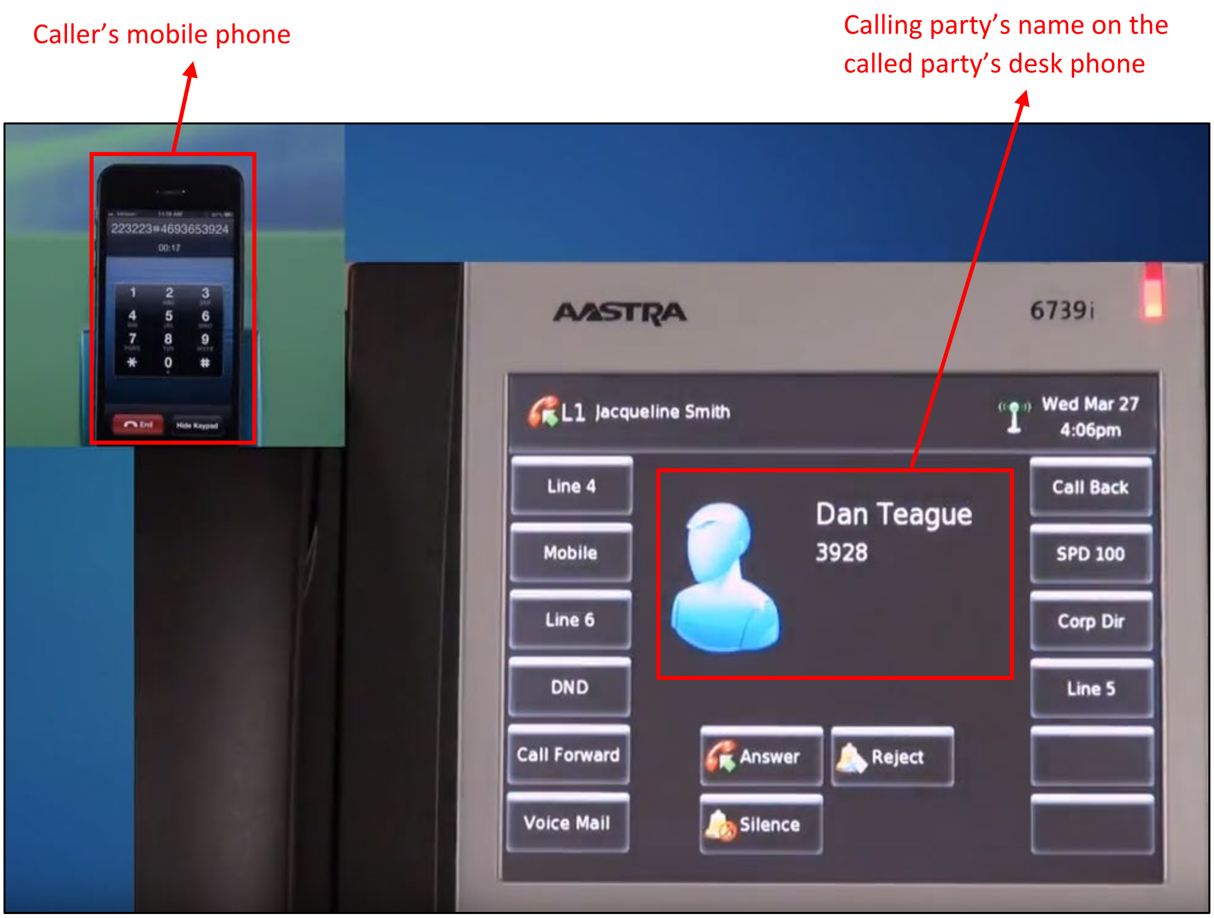
The basic service works as follows:

- If the name information is already present in the incoming call setup message, then the external database is not accessed.
- If the name information is not available, Clearspan sends a request to the external database.

The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

138. The methods practiced by Mitel's use of the accused products include causing the LR to provide the information in a response to the query. For example, the database (LR) present in the application server of Clearspan server provides the caller name information associated with the caller number. Further, the caller name (for example: Dan Teague) information is presented to the called party.

Clearspan Server Components

Within the BladeCenter chassis, servers are populated by function and capacity requirements. The following server types are populated within the chassis:

- Application Server (AS) – provides call processing, feature logic, user management, call detail records and service management. Deployed as Active / Standby.

(Source: http://edocs.mitel.com/UG/AASTRA/Archive/Clearspan/274007.pdf)

A user who has the Calling Name Retrieval service assigned and enabled can make Caller ID with NAMe (CNAM) database queries. However, if that user does not have the Calling Name Delivery service enabled, then the Application Server does not deliver the calling name to the user. Note that such service configuration may have value, since the Application Server can deliver the calling name to Attendant Console users who are monitoring that user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R21/281106.pdf)

CALLING NAME RETRIEVAL

The Calling Name Retrieval service allows Clearspan to provide a caller's name to a user by retrieving the calling name from a Public Switched Telephone Network (PSTN)-hosted database through a Signaling System 7 (SS7)-enabled network element such as a softswitch.

DESCRIPTION

This service provides a SIP subscription-based method or a Simple Object Access Protocol/ Extensible Markup Language (SOAP/XML) query method of retrieving calling name information from an external database on a per-call basis; this function is analogous to the GR-1188 Transactional Capabilities Application Part (TCAP) terminating query.

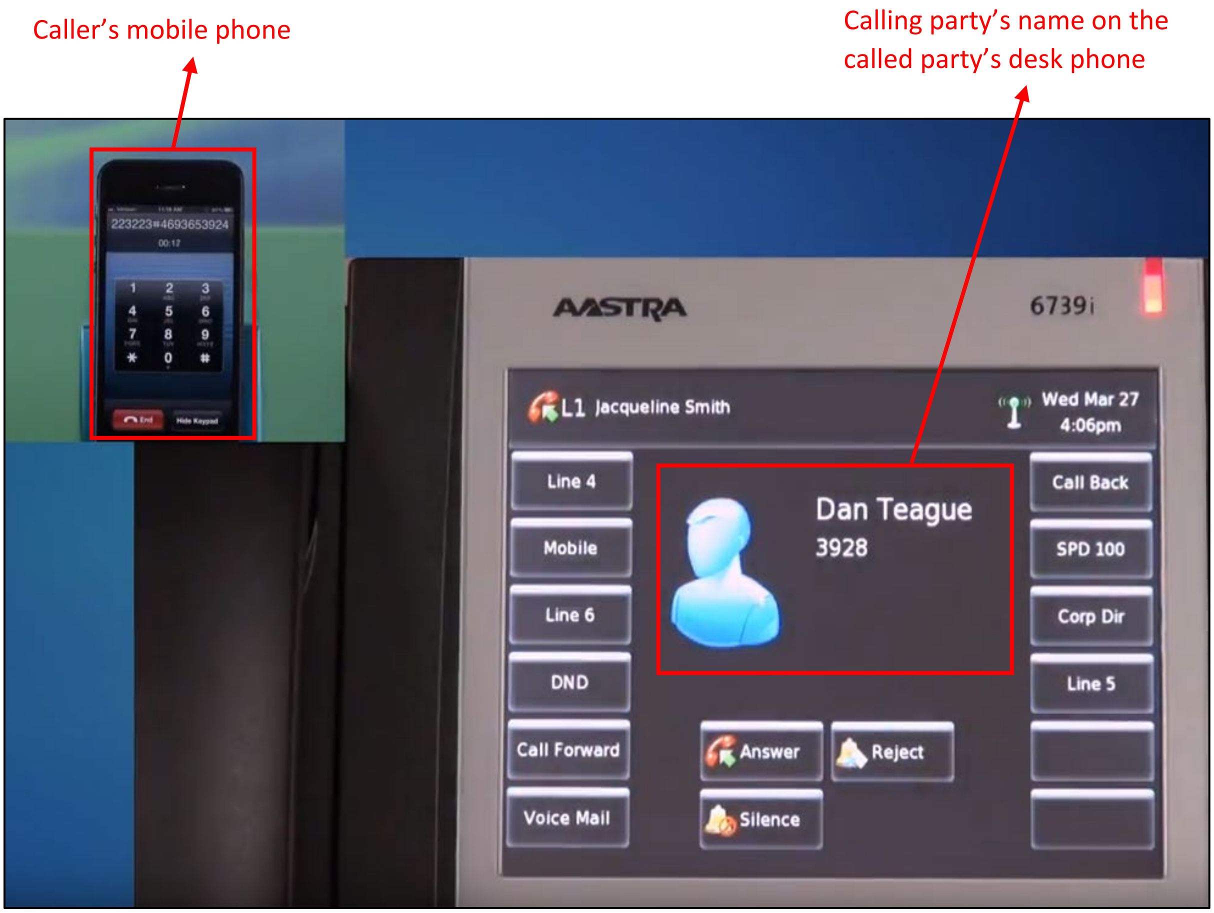
The basic service works as follows:

- If the name information is already present in the incoming call setup message, then the external database is not accessed.
- If the name information is not available, Clearspan sends a request to the external database.

The query contains the caller's number, which allows the external database to look up the caller's name. When Clearspan receives a response from the external database, the caller's name information is extracted from the message and is relayed to the user.

(Source:

http://edocs.mitel.com/UG/AASTRA/TechDocs/Clearspan/Clearspan_R22/282408%20CS%20Service%20Guide%20R22.pdf)



(Source: screenshot of video available at https://www.youtube.com/watch?v=h5jFvtueVeE)

139. Far North Patents only asserts method claims from the '588 Patent.

140. Mitel has had knowledge of the '588 Patent at least as of the date when it was notified of the filing of this action.

141. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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.

142. 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

143. Mitel 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. Mitel has induced the end-users, Mitel'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.

144. Mitel 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.

145. Such steps by Mitel 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.

146. Mitel 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.

147. Mitel was and is aware that the normal and customary use of the accused products by Mitel’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. Mitel’s inducement is ongoing.

148. Mitel 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.

149. Mitel has at least a significant role in placing the accused products in the stream of commerce in Texas and elsewhere in the United States.

150. Mitel 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.

151. Mitel directs or controls the sale of the accused products into established United States distribution channels, including sales to nationwide retailers.

152. Mitel’s established United States distribution channels include one or more United States based affiliates (e.g., at least Mitel Networks, Inc., Mitel Technologies, Inc., Mitel Business Systems, Inc., Mitel Cloud Services, Inc., and Mitel Communications Inc.).

153. Mitel directs or controls the sale of the accused products in nationwide retailers such as CDW, including for sale in Texas and elsewhere in the United States, and expects and intends that the accused products will be so sold.

154. Mitel 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.

155. Such steps by Mitel 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.

156. Mitel 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.

157. Mitel performed such steps in order to profit from the eventual sale of the accused products in the United States.

158. Mitel's inducement is ongoing.

159. Mitel 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. Mitel 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.

160. 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.

161. 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.

162. 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.

163. Mitel's contributory infringement is ongoing.

164. Furthermore, Mitel 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).

165. Mitel'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 Mitel.

166. Mitel 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.

167. Mitel'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.

168. Mitel encouraged its customers' infringement.

169. Mitel'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.

170. Far North Patents has been damaged as a result of the infringing conduct by Mitel alleged above. Thus, Mitel 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 Mitel, 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 Mitel and/or all others acting in concert therewith;

b. A permanent injunction enjoining Mitel 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 Mitel account for and pay to Far North Patents all damages to and costs incurred by Far North Patents because of Mitel’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 Mitel’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,

/s/ *Zachariah S. Harrington*

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