IN THE UNITED STATES DISTRICT COURT
FOR THE WESTERN DISTRICT OF TEXAS
WACO DIVISION

THINKLOGIX, LLC, Plaintiff, v. VTECH HOLDINGS LTD. and VTECH COMMUNICATIONS, INC., Defendants.	Case No. 6:24-cv-00206 **Jury Trial Demanded**

COMPLAINT FOR PATENT INFRINGEMENT

Plaintiff ThinkLogix LLC, ("ThinkLogix") files this Complaint against VTech Holdings Ltd. and VTech Communications, Inc., (unless otherwise stated, hereinafter collectively referred to as "VTech" or "Defendants") for infringement of United States Patent Nos. 6,920,373; 8,599,835; 9,231,994; 9,906,573; 7,184,524; and 7,136,392; (the "Patents-in-Suit"), and alleges as follows:

NATURE OF THE ACTION

1. This is an action for patent infringement arising under the patent laws of the United States, 35 U.S.C. §§ 1 *et seq.*

THE PARTIES

2. ThinkLogix LLC is a Texas limited liability company with its principal place of business at 17350 State Highway 249 STE 220, Houston, Texas 77064-1132, USA.

3. On information and belief, defendant VTech Holdings Ltd. is a foreign corporation of Hong Kong, having a principal place of business at 23rd Floor, Tai Ping Industrial Centre, Block 1, 57 Ting Kok Road, Tai Po, New Territories, Hong Kong.

4. On information and belief, defendant VTech Communications, Inc., is a corporation organized under the laws of Oregon, having a principal place of business 9020 South West Washington Square, Road, Suite 555, Portland, Oregon 97223.

5. On information and belief, defendant VTech Communications, Inc., may be served with process via its registered agent, Unisearch Inc., located at 823 Congress Avenue, Suite P-4, Austin, Texas 78701.

6. On information and belief, VTech Communications, Inc has been authorized to do business in the State of Texas and the Western District of Texas since at least April 6, 2006, under Texas SoS File Number 0800638388.

JURISDICTION AND VENUE

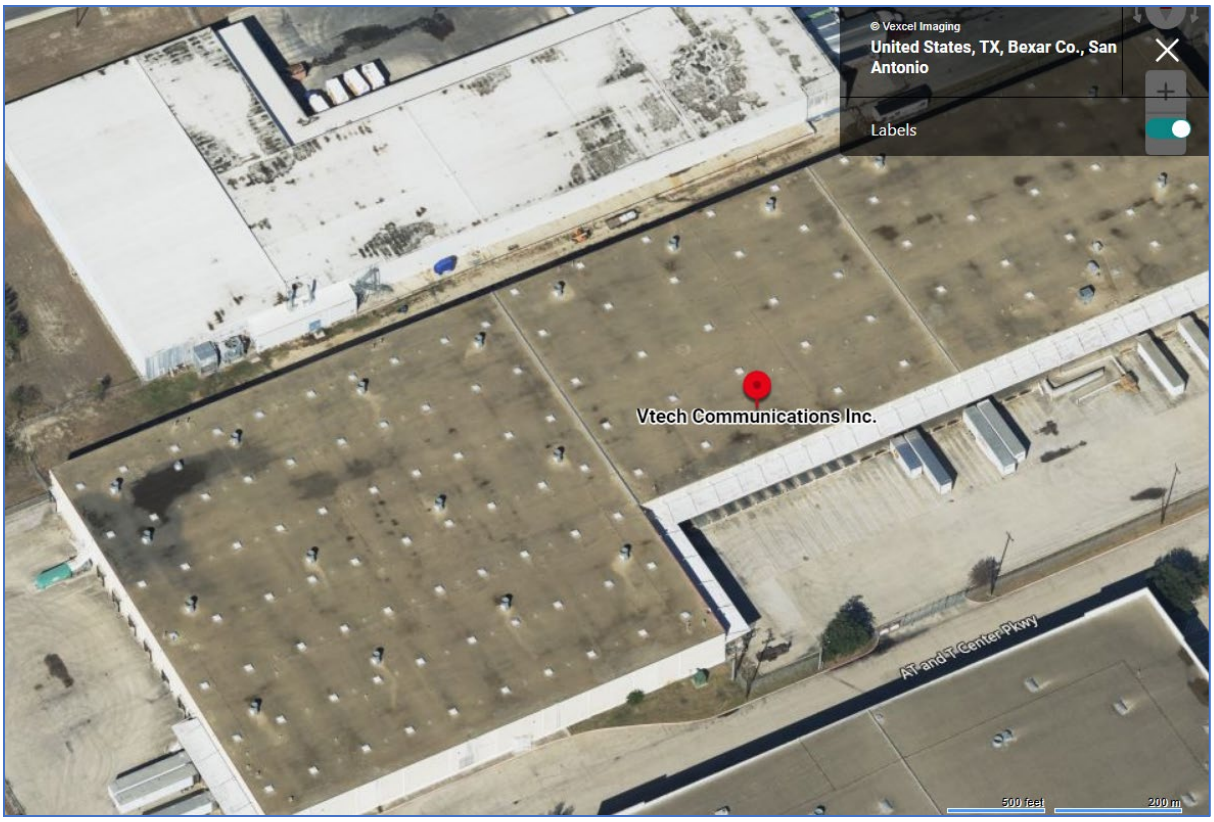
7. On information and belief, VTech provides cloud-based services related to its security camera monitoring platforms to its customers, including but not limited to those customers in this Judicial District.

8. On information and belief, VTech also provides mobile and online call conferencing services associated with its call conferencing products to its customers, including but not limited to those customers in this Judicial District.

9. The Patents-in-Suit cover VTech's products, services, and methods related to conference and camera/video systems, which are designed, developed, manufactured, distributed, sold, offered for sale, and used by VTech and/or their customers, consumers, and clients, including but not limited to those customers, consumers, and clients residing in the State of Texas and this Judicial District.

10. On information and belief, VTech, on its own and/or via its divisions, subsidiaries, partners, and affiliates maintain a corporate and commercial presence in the United States, including in the State of Texas and this judicial district, via at least 1) VTech's physical locations, established throughout Texas, including this District; 2) VTech's online presence that provides to consumers access to VTech's Products and Services, including those identified as infringing herein; and 3) consumers and clients of VTech who utilize VTech services, at physical and online sites.

11. VTech, on its own and/or via alter egos, agents, divisions, subsidiaries, partners, and affiliates maintains a regular and established place of business in this District located at 1143 AT&T Center Parkway, San Antonio, Texas 78219, do business, including committing infringing acts, in the U.S., the State of Texas, and in this Judicial District.



12. On information and belief, VTech has made, used, offered to sell and/or sold products and services, including the following specifically accused products and services: (1) MyVTech Baby Pro app;[1] (2) Baby Monitor;[2] (3) VTech Snom's Desk Telephone D717, D725 and similar products;[3] (4) VTech's '2-Line Color SIP Cordless Accessory Handset and similar products;[4] (5)VTech's KidiBuzz G2

[1] *See e.g.*, https://apps.apple.com/us/app/myvtech-baby-pro/id1542759321 and https://play.google.com/store/apps/details?id=com.cams.vtech.mvb.pro.

[2] *See e.g.*, https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.

[3] *See e.g.*, https://www.snomamericas.com/en/pd/ip-phones/desk-phones/d7xx-series-next-gen/d717 and https://www.snomamericas.com/en/pd/d725.

[4] *See e.g.*, https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

[5] *See e.g.*, https://www.vtechkids.com/kidibuzzsafety.

Smart Device;[5] (6) current or legacy products or services, which use, or have used, one or more of the foregoing products and services as a component product or component service; (7) combinations of products and/or services comprising, in whole or in part, two or more of the foregoing products and services; (8) hardware and software components comprising the VTech Products and Services and/or that enable the VTech Products and Services to operate, including but not limited to cordless phones, desk phones, webcams, baby-monitors, smart devices, servers, server hardware, server software, website software, webservers, client-side software, mobile software, mobile app software, and browser executable software; and (9) all other current or legacy products and services imported, made, used, sold, or offered for sale by VTech that operate, or have operated in a substantially similar manner as the above-listed products and services. (As used herein, one or more of the forgoing products and services are individually and collectively referred to as the accused "VTech Products and Services"). On information and belief, one or more of the VTech Products and Services infringes (literally and/or under the doctrine of equivalents) at least one claim of each of the patents-in-suit.

13. This Court has personal jurisdiction over VTech, at least, because it committed and continues to commit acts of infringement in this judicial district in violation of 35 U.S.C. § 271. In particular, on information and belief, VTech has made, used, offered to sell access to, and/or sold access to the accused VTech Products and Services in the Western District of Texas, and has made, used, offered to sell access to, and/or sold access to the VTech System in the Western District of Texas.

14. On information and belief, VTech is subject to the Court's jurisdiction because it regularly conducts and solicits business, or otherwise engages in other persistent courses of conduct in this judicial district, and/or derives substantial revenue from the use, sale, and distribution of goods and services, including but not limited to the accused VTech Products and Services provide to individuals and businesses in the Western District of Texas.

15. On information and belief, VTech directly infringes the patents-in-suit in Texas, including specifically in the Western District of Texas, at least, by making, using, offering to sell access to, and/or selling access to the accused VTech Products and Services in the Western District of Texas, and its making, use, offering to sell access to, and/or selling access to the VTech System in the Western District of Texas.

16. On information and belief, one or more of the accused VTech Products and Services and/or the VTech System are made, used, sold and/or offered for sale by VTech, its subsidiaries and/or agents, in the Western District of Texas.

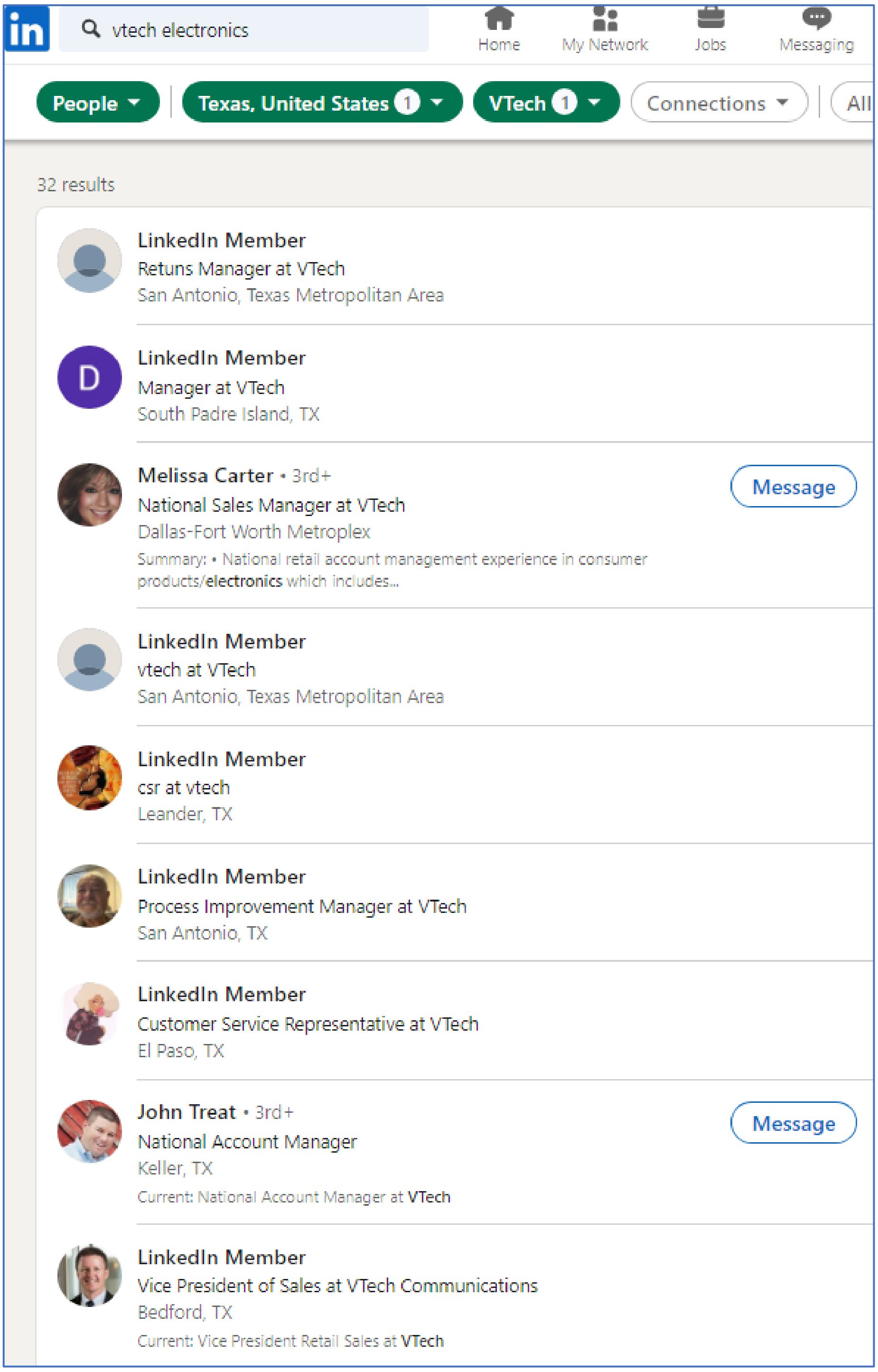
17. On information and belief, VTech's customers located in the Western District of Texas have obtained access to and used the accused VTech Products and Services and/or the VTech System while located in the Western District of Texas.

18. The Court has personal jurisdiction over VTech, at least, because it has continuous business contacts in the State of Texas and in the Western District of Texas and VTech has engaged in business activities including transacting business in the Western District of Texas and purposefully directing its business activities, including the sale or offer for sale of the VTech Products and Services to the Western District of Texas to induce, aid, abet, and/or contribute to the infringement of third

parties in the Western District of Texas, including without limitation the direct infringement of VTech's customers located in the Western District of Texas through the use of VTech Products and Services and the VTech System, while they are located within the Western District of Texas.

19. This Court has personal jurisdiction over VTech because, *inter alia*, VTech, on information and belief: (1) has committed acts of patent infringement in this Western District of Texas; (2) maintains a regular and established place of business in the Western District of Texas; (3) has substantial, continuous, and systematic contacts with this State and the Western District of Texas; (4) owns, manages, and operates facilities in this State and the Western District of Texas; (5) enjoys substantial income from its operations and sales in this State and the Western District of Texas; (6) employs Texas residents in this State and the Western District of Texas, and (7) solicits business using the VTech Products and Services and VTech System in this State and the Western District of Texas.

20. On information and belief, VTech supports and markets the VTech Products and Services and VTech System in the State of Texas and this Judicial District, to customers and potential customers, who reside in the State of Texas and in this judicial district through various means, including through the efforts of its agents and/or employees who reside in the Western District of Texas.



21. Venue is proper pursuant to 28 U.S.C. §§ 1391(b), (c), (d) and/or 1400(b) and the Federal Circuit's decision in *In re Monolithic Power Sys.*, 50 F.4th 157 (Fed. Cir. 2022), at least because VTech has multiple employees based out of this district as listed on LinkedIn, has transacted business in this district, and has directly

committed acts of patent infringement in this district. Venue is also proper as to Defendant VTech Holdings Ltd, which is a foreign corporation, under 28 U.S.C. § 1391(c)(3) that provides "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."

<u>VTech's Prior Knowledge of the Patents-in-Suit</u>

22. On or about October 31, 2023, counsel for ThinkLogix sent a letter (the "October 2023 Letter") via FedEx to Dr. Allan Wong Chi Yun, in his role as Chairman and Group Chief Executive Officer of VTech, as well as to Mr. Kenton Erwin in his role as General Counsel of VTech. The October 2023 Letter was sent to VTech, along with a draft of the instant Complaint, for the express purpose of acquainting VTech with ThinkLogix's patent portfolio of at least the '373; '835; '994; and '573 Patents, prior to any enforcement action relating to its past and future use, related to VTech's System, and invited VTech to participate in discussions regarding a license to allow its continued use of ThinkLogix's patented technologies.

23. As of the date of the filing of the instant Complaint, VTech has not responded to the October 2023 Letter.

24. On information and belief, at all times from October 2023 to the filing of the instant Complaint, Mr. Erwin has held the office of General Counsel of VTech.

<u>United States Patent No. 6,920,373</u>

25. On July 19, 2005, the USPTO duly and legally issued United States Patent No. 6,920,373 ("the '373 patent") entitled "Synchronization and task control of real-time internet based super-media" to inventor Ning Xi, and Imad Elhajj.

26. The ’373 patent is presumed valid under 35 U.S.C. § 282.

27. ThinkLogix owns all rights, title, and interest in the ’373 patent.

28. ThinkLogix has not granted VTech an approval, an authorization, or a license to the rights under the ’373 patent.

29. The ’373 patent relates to, among other things, the control of devices, and more particularly to super-media enhanced control of devices over the Internet.

30. The claimed inventions(s) of the ’373 patent sought to solve problems with, and improve upon, attempts to control devices over the Internet with force feedback.

31. The inventions claimed in the ’373 patent overcome the limitations in the prior art to effectively control a device remotely in a closed-loop control system. The ’373 patent discloses a closed-loop control system that includes a device that is connected to the distributed communications system and that generates super-media feedback signals. A computer is connected to the distributed communications system and includes a controller that provides super-media feedback. The computer generates and transmits command signals using the controller to the device and outputs the super-media feedback signals to the controller. The closed-loop control system is event-based to ensure stability and synchronization of the closed-loop system.

United States Patent No. 8,599,835

32. On December 3, 2013, the USPTO duly and legally issued United States Patent No. 8,599,835 (“the ’835 patent”) entitled “Streaming media” to inventors Martti Mela, and Pekka Pessi.

33. The ’835 patent is presumed valid under 35 U.S.C. § 282.

34. ThinkLogix owns all rights, title, and interest in the ’835 patent.

35. ThinkLogix has not granted VTech an approval, an authorization, or a license to the rights under the ’835 patent.

36. The ’835 patent relates to, among other things, a novel method of multimedia streaming based on the SIP protocol.

37. The claimed invention(s) of the ’835 patent sought to solve problems with, and improve upon, existing streaming media solutions. For example, the ’835 patent states:

> According to a first aspect of the present invention there is provided a method for providing an SIP session between a first and a second entity, said method comprising the steps of establishing a SIP session between the first and second entity; transmitting at least one media stream between the first entity and the second entity and controlling at least one of transmission, storage and play back of the at least one media stream at said first and/or second entity.
>
> According to a second aspect of the present invention there is provided a method of providing an SIP session between a first and a second entity, and a RTSP session between said second entity and a third entity, said method comprising the steps of establishing a SIP session between the first and second entity; establishing an RTSP session between said second entity and said third mapping between SIP and RSTP information to allow data to pass from one of said first and third entities to the other of said first and third entities.
>
> According to a third aspect of the present invention, there is provided a method of providing a SIP session between a first entity and a second entity, said method comprising the steps of establishing a SIP session between the first and second entities; providing at least one media stream to one of said first and second entities from the other of said entities; and updating said at least one media stream during said session.
>
> According to a fourth aspect of the present invention, there is provided a communications system comprising a first and second entity, said first and second entity being operable to establish a SIP session between the first and second entity, one of said entities being arranged to transmit at least one media streams to the other entity, said system being such that at least one of

> transmission, storage and play back of the media streams is controlled at least one of said entities.
>
> According to a fifth aspect of the present invention, there is provided a communications system comprising a first and second entity, said first and second entity being operable to establish a SIP session between the first and second entity, said first entity being arrange to retrieve a description of a session, said first and second entities being arranged to establish a session therebetween, one of the entities being arranged to send at least one media stream to the other entity and said other entity being arranged to control the play back of the media streams.
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> According to a sixth aspect of the present invention, there is provided a communications system comprising a first and a second entity, said first and second entity being operable to establish a SIP session between the first and second entity, one of said entities being arranged to transmit at least one media stream to the other entity, wherein said at least one of said media streams can be sent at one of a plurality of different playback rates.
>
> According to a seventh aspect of the present invention, there is provide a communications system comprising a first, second and third entity wherein the first and second entities are arranged to establish a SIP session therebetween, and the second and third entities are arrange to establish a RTSP session therebetween, said second entity being arranged to map between SIP and RSTP information to allow data to pass from one of said first and third entities to the other of said first and third entities.
>
> According to an eighth aspect of the present invention, there is provided a communications system comprising a first and a second entity, said first and second entity being operable to establish a SIP session between the first and second entity, one of said entities being arranged to transmit at least one media stream to the other entity, wherein said at least one media stream us updatable during said session.
>
> According to a ninth aspect of the present invention, there is provided a gateway for use in a communications system, wherein said gateway is arranged to establish a SIP session with one entity and a RTSP session with a further entity, said gateway being arranged to map between SIP and RSTP information to allow data to pass from one of said first and third entities to the other of said first and third entities.

See e.g., ’835 Specification col. 2, l. 5 – col. 3, ll. 1- 10.

38. The inventions claimed in the ’835 patent overcome the limitations of prior art to enhance the user experience in streaming media, including but not

limited to: (1) the ability to receive a request to establish a media session for streaming media from a media source to a media sink, which allows for the initiation of a media streaming session; (2) the capacity to send a response to the request that includes a session description and an indication of a maximum bit rate for the media session, which provides a mechanism for controlling the quality and bandwidth usage of the media stream; (3) the feature of receiving a media stream from the media source at a bit rate that does not exceed the maximum bit rate, which ensures efficient use of network resources and prevents network congestion; (4) the inclusion of a method for dynamically adjusting the maximum bit rate based on network conditions, which provides flexibility in managing network resources and can lead to improved quality of service; and (5) the ability to provide a smoother streaming experience to the user by managing the bit rate of the media stream, which can reduce issues such as buffering or poor video quality.

United States Patent No. 9,231,994

39. On January 5, 2016, the USPTO duly and legally issued United States Patent No. 9,231,994 ("the '994 patent") entitled "Streaming media" to inventors Martti Mela and Pekka Pessi.

40. The '994 patent is presumed valid under 35 U.S.C. § 282.

41. ThinkLogix owns all rights, title, and interest in the '994 patent.

42. ThinkLogix has not granted VTech approval, an authorization, or a license to the rights under the '994 patent.

43. The '994 patent relates to, among other things, multimedia streaming based on SIP protocol.

44. The specification of the ’994 patent is the same as the ’835 patent specification and addresses and solves the problems recited above and described in the ’835 patent specification.

United States Patent No. 9,906,573

45. On February 27, 2018, the USPTO duly and legally issued United States Patent No. 9,906,573 (“the ’573 patent”) entitled “Streaming media” to inventors Martti Mela and Pekka Pessi.

46. The ’573 patent is presumed valid under 35 U.S.C. § 282.

47. ThinkLogix owns all rights, title, and interest in the ’573 patent.

48. ThinkLogix has not granted VTech an approval, an authorization, or a license to the rights under the ’573 patent.

49. The ’573 patent relates to, among other things, multimedia streaming based on SIP protocol.

50. The specification of the ’573 patent is the same as the ’835 patent specification and addresses and solves the problems recited above and described in the ’835 patent specification.

United States Patent No. 7,184,524

51. On February 27, 2007, the USPTO duly and legally issued United States Patent No. 7,184,524 (“the ’524 patent”) entitled “Rule Based Real-Time Communication System” to inventors Charles J. Digate, Christopher F. Herot, Tonytip Ketudat, and Alexis M. Kopikis.

52. The ’524 patent is presumed valid under 35 U.S.C. § 282.

53. ThinkLogix owns all rights, title, and interest in the ’524 patent.

54. ThinkLogix has not granted VTech an approval, an authorization, or a license to the rights under the ’524 patent.

55. The ’524 patent relates to, among other things, real-time communication, and, more specifically, to a rules-based real-time communication system that facilitates conference scheduling, information distribution and communication among a plurality of users.

56. The claimed invention(s) of the ’524 patent sought to solve problems with, and improve upon, existing real-time communication solutions. For example, the ’524 patent states:

> A rules based real-time messaging system for groups of users is disclosed, in which an availability status may be maintained in association with each user. A number of client systems are communicably coupled to a real-time messaging server via a network. The real-time messaging server applies a set of rules to the availability status of users, as well as other attributes associated with users, in order to facilitate effective real-time message passing between users. As described herein, the availability status of a user reflects what is generally referred to as the online presence of that user.
>
> The real-time messaging server includes a number of rules and a rules engine for controlling delivery of messages to users, and for controlling how user availability is communicated among users. Rules stored in the rules database, or the rules engine itself, include various types of rules, including “when and "if-then type rules. Based on the rules stored on the real-time messaging server, the rules engine determines the state of relevant conditions, such as the availability status of specific users, and responds to the occurrence of relevant real-time events, such as a user logging-on or logging-off, in order to control the delivery of various messages and/or performance of resulting actions.
>
> The disclosed system facilitates real-time group interaction by monitoring events and testing for conditions, and taking appropriate actions. Through the rules and rules engine in the real-time messaging server, the disclosed system enables users to control their availability to other users, and how they are accessed for real-time activities Such as online meetings or teleconferences. As a result, the disclosed real-time messaging system is particularly Suited for setting up real-time activities among key individuals. The disclosed system advantageously employs online presence indicators to conveniently convene a conference immediately when possible, to convene a conference as soon as

> possible if it is not feasible to convene the conference immediately, and/or to schedule a conference at a predetermined time in the future, as well as assuring the participation of needed contributors in a conference or other real-time activity. Moreover, while the disclosed system is described herein with reference to various embodiments and examples of operation in which convening a meeting is used as an example of a real-time activity or action, the real-time activities or actions provided by or in connection with the disclosed system are not limited to meetings between users, and may additionally or alternatively include chat sessions, shared whiteboards, remote presentations, audio conferences, video conferences, and/or any combination of these or other forms of communication between users.
>
> Other features, aspects and advantages of the presently disclosed system and method will be apparent from the detailed description of the invention that follows.

See e.g., '524 Specification col. 2, l. 33 – col. 3, ll. 1- 18.

57. The inventions claimed in the '524 patent overcome the limitations of prior art to enhance the user experience in real-time communications, including but not limited to: (1) the ability to have a real-time messaging system designed to facilitate real-time convening of conferences and business communication; (2) the availability status of a user reflecting what is generally referred to as the online presence of a user; (3) facilitating real-time group interaction by monitoring events, testing conditions and taking appropriate action; and (4) additionally or alternatively including chat session, shared whiteboards, remote presentations, audio conferences, video conferences, and/or any combination of these forms of communication between users.

<u>United States Patent No. 7,136,392</u>

58. On November 14, 2006, the USPTO duly and legally issued United States Patent No. 7,136,392 ("the '392 patent") entitled "System and Method for

Ordering Data Message Having Differing Levels of Priority for Transmission Over a Shared Communication Channel" to inventor Maarten Menzo Wentink.

59. On September 9, 2015, the Patent Trial and Appeal Board at the USPTO duly and legally ordered that several claims including Claim 4 of the '392 to be patentable.

60. The '392 patent is presumed valid under 35 U.S.C. § 282.

61. ThinkLogix owns all rights, title, and interest in the '392 patent.

62. ThinkLogix has not granted VTech an approval, an authorization, or a license to the rights under the '392 patent.

63. The '392 patent relates to, among other things, method and apparatus for classifying and ordering data messages prior to their transmission over a shared communication channel.

64. The claimed invention(s) of the '392 patent sought to solve problems with, and improve upon, media control in communication networks. For example, the '392 patent states:

> A method according to an illustrative embodiment of the invention comprises directing, to a first output queue at a first station of a communication network, data message units that are to be transmitted over a communication medium and that have a first traffic classification. The method further comprises directing to a second output queue at the first station, data message units that are to be transmitted over the communication medium and that have a second traffic classification. Each queue senses the communication medium for an opportunity to transmit data message units according to the aforementioned set of rules. An attempt is made to retransmit, after a respective interval defined differently for each corresponding traffic classification, any message data unit transmitted by the first station that destructively interferes with a message data unit transmitted by another station over the communication medium, or with other causes of interference.
>
> If the first and second output queues at the first station each contain data message units that are scheduled to be transmitted during the same sensed

> transmit opportunity, they could benefit from that prior knowledge and, by way of illustrative example, at least one message data unit from the first output queue can be preferentially transmitted over a message data unit from the second queue. For purposes of this specification, preferential treatment encompasses one or more measures that guarantee—for streams of higher priority data message units—an average transmission rate that is higher and/or a maximum delay before transmission that is lower than that which is guaranteed to streams of lower priority data message units. Such measures include, by way of illustrative example, deferring transmission of (i.e., "preempting') a lower priority message data unit from the second output queue to ensure that the higher priority message data unit can be transmitted during a specific transmit opportunity.
>
> According to the present invention, however, an effort is made to fairly allocate transmission opportunities among every queue having data message units of the same traffic classification to transmit-regardless of where each queue happens to be among the stations of the network. Specifically, each station is configured so that a queue containing higher priority data message units has no more impact on the scheduling order of a local queue (i.e., a queue in the same station) than it does on the scheduling order of any external queue (a queue at any other station) that contains data message units of the same, lower priority level. To this end, in the illustrative embodiment of the invention, at least one attempt is made to transmit a data message unit, from the second output queue, that has been pre-empted by a higher priority message data unit from the first output queue, as if that pre-empted message data unit had already been transmitted and had experienced destructive interference. As such, the probability that a message data unit of a given priority level will be transmitted during a Subsequent transmission opportunity is Substantially the same regardless of whether it was actually transmitted and destructively interfered with a message data unit from another station or it was internally pre-empted by a higher priority message data unit before it could be transmitted.

See e.g., ’392 Specification col. 2, l. 24 – col. 3, ll. 1- 15.

65. The inventions claimed in the ’392 patent overcome the limitations of prior art to enhance media control in communication networks, including but not limited to: (1) fairly allocating transmission opportunities between all queues containing data messages of the same priority level; (2) directing message data units with first traffic classification to an output queue at a first station of a communication network; (3) directing message data units with a second traffic

classification to an output queue at a first station; (4) sensing a communication medium for an opportunity to transmit message data units without interference from message data units transmitted by a second station, according to sets of rules that vary by traffic classification yet are common to the first station and second station; (5) attempting to retransmit, after a respective interval defined differently by a set of rules, any message data unit transmitted over the communication medium by a station that collides with a message data unit transmitted by another station; (6) attempting to initially transmit a first message data unit from the second output queue of the first station, in accordance with the set of rules corresponding to a traffic classification thereof, as if an unsuccessful attempt to transmit the first message data unit had already been made during a previous transmission opportunity; and (7) attempting to transmit the first message data unit after an interval specified by the set of rules corresponding to the traffic classification of the second queue.

CLAIMS FOR RELIEF

Count I – Infringement of United States Patent No. 6,920,373

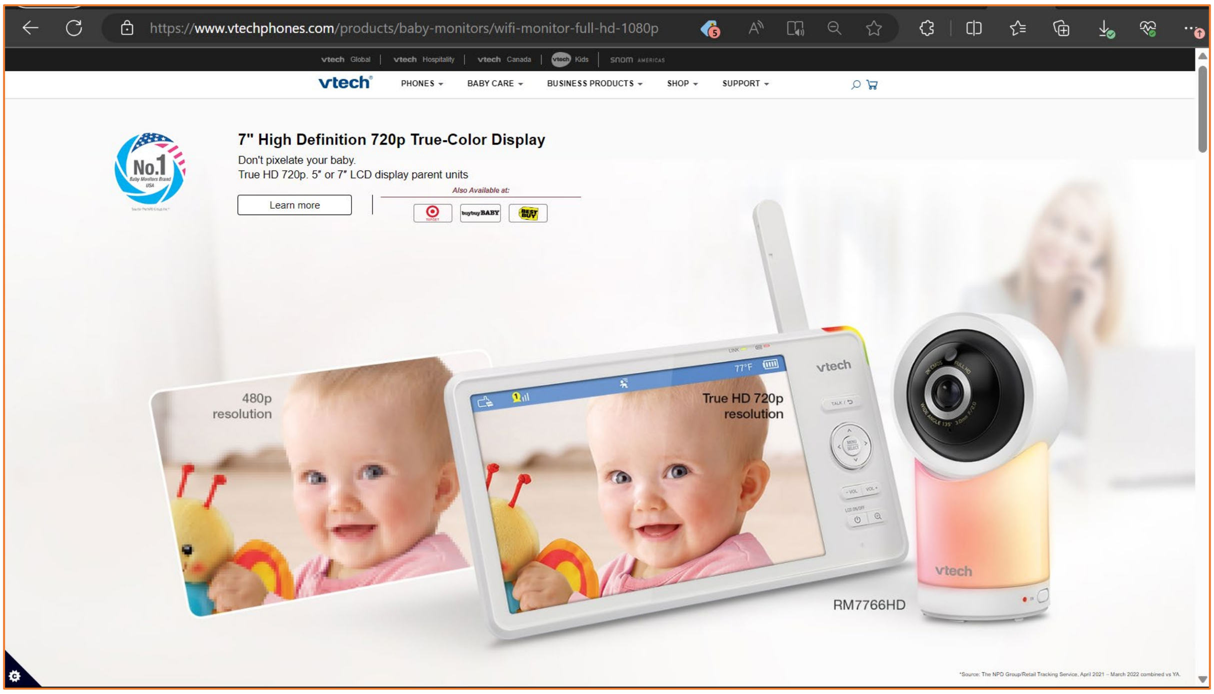
66. ThinkLogix repeats, realleges, and incorporates by reference, as if fully set forth here, the allegations of the preceding paragraphs above.

67. On information and belief, VTech is in violation 35 U.S. C. § 271(a) with respect to one or more claims of the ’373 patent.

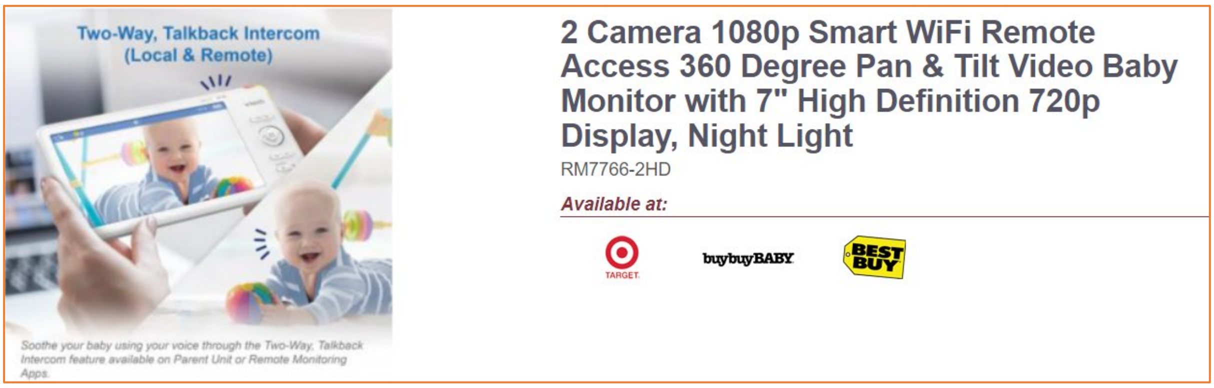
68. On information and belief, VTech (or those acting on its behalf) has made, used, offered to sell, sold, and/or imported one or more components of the

VTech Products and Services that infringes (literally and/or under the doctrine of equivalents) at least claim 1 of the '373 patent.

69. On information and belief, one or more components of the VTech Products and Services comprises a closed-loop control system, (*e.g.*, a baby monitor) that remotely controls a device (*e.g.*, a baby camera unit) over the Internet.



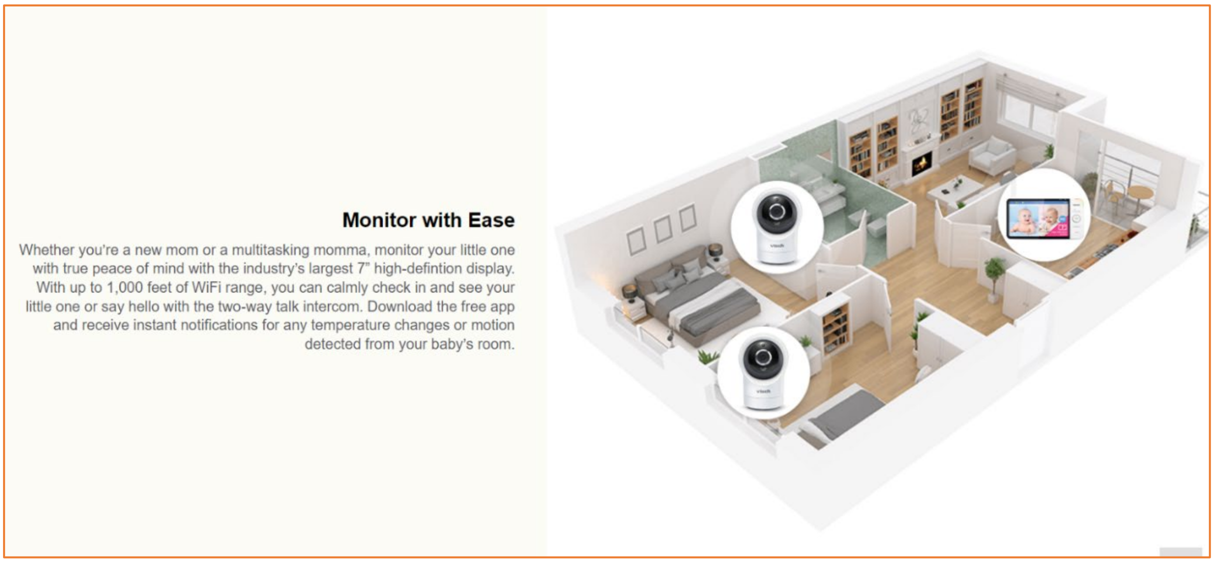
See e.g., https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.



See e.g., https://www.VTechphones.com/store/pd/4710/RM7766-2HD-2-Camera-1080p-Smart-WiFi-Remote-Access-360-Degree-Pan-Tilt-Video-Baby-Monitor-with-7-High-Definition-720p-Display-Night-Light.

Free live remote access via smartphone and tablet
Free live remote access via smartphone and tablet to remotely listen, talk-back and view on multiple devices, even outside the home.

See e.g., https://www.VTechphones.com/store/pd/4710/RM7766-2HD-2-Camera-1080p-Smart-WiFi-Remote-Access-360-Degree-Pan-Tilt-Video-Baby-Monitor-with-7-High-Definition-720p-Display-Night-Light.



See e.g., https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.

What does HD Video Monitor do?

VTech 7" Smart Wi-Fi 1080p Pan & Tilt Monitor allows you to stay close to your baby when you are on the move or away. This monitor is Wi-Fi enabled, using your home wireless network to stream live video and audio through the 1080p baby unit and 7-inch color screen parent unit. It allows you to maintain a constant connection to your baby.

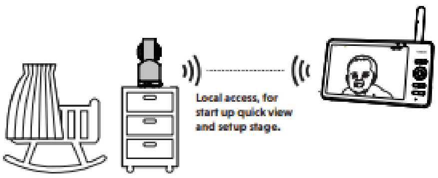
You can also monitor from your smartphone and mobile tablet.
VTech 7" Smart Wi-Fi 1080p Pan & Tilt Monitor uses **MyVTech Baby Pro** app to let you instantly see and hear what's happening with your baby from anywhere in the world. The app can be downloaded from the App Store or the Google Play™ Store.

See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

How the system works

Direct mode

Your baby monitor parent unit and baby unit connect to each other in Direct mode by default, which allows you to immediately stream video out of the box.
It is NOT recommended to continue using Direct Mode connection after initial installation.



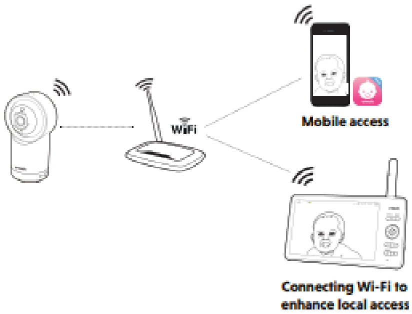
TIP

You may experience reduced signal strength if your parent unit and baby unit are far apart, or there are obstructing factors, such as internal walls, between them. You have to set up **Router mode** to connect the units via your home Wi-Fi network to improve the connection between your parent unit and baby unit.

Router mode

You are required to set up your monitor in **Router mode** to enjoy live video streaming via your parent unit with an enhanced connection. **Router mode** is the recommended mode for stable connectivity.

Router mode uses your home Wi-Fi network to connect your video monitor. It supports live video streaming through the parent unit and smartphone app.

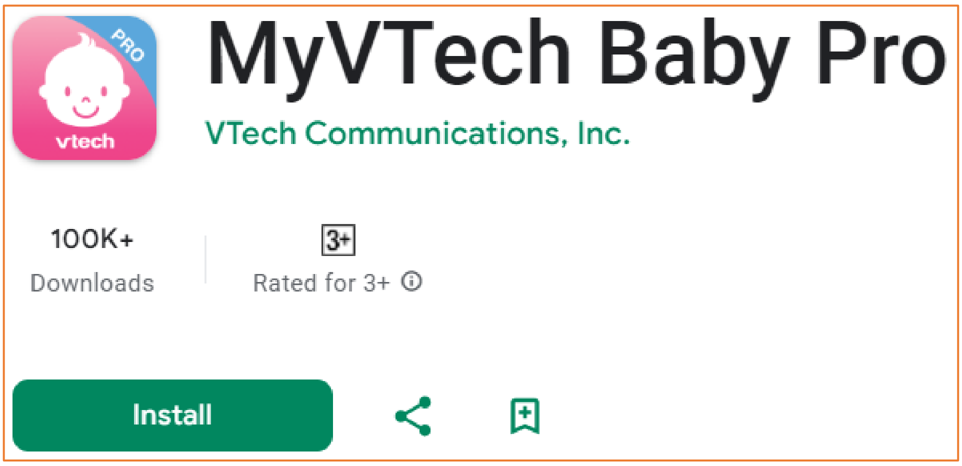


If you want to stream live video remotely with your mobile device, download and install the **MyVTech Baby Pro** mobile app. With the mobile app, your Wi-Fi router (not included) provides Internet connectivity to your HD video monitor system. It serves as a communicating channel between your baby unit and mobile device, allowing you to monitor and/or control your baby unit wherever you are. For more details, see **Download app for mobile access** section in this quick start guide.

NOTE

- When both parent unit and app are viewing the camera simultaneously, the video quality will be optimised to deliver the best possible viewing result.

See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

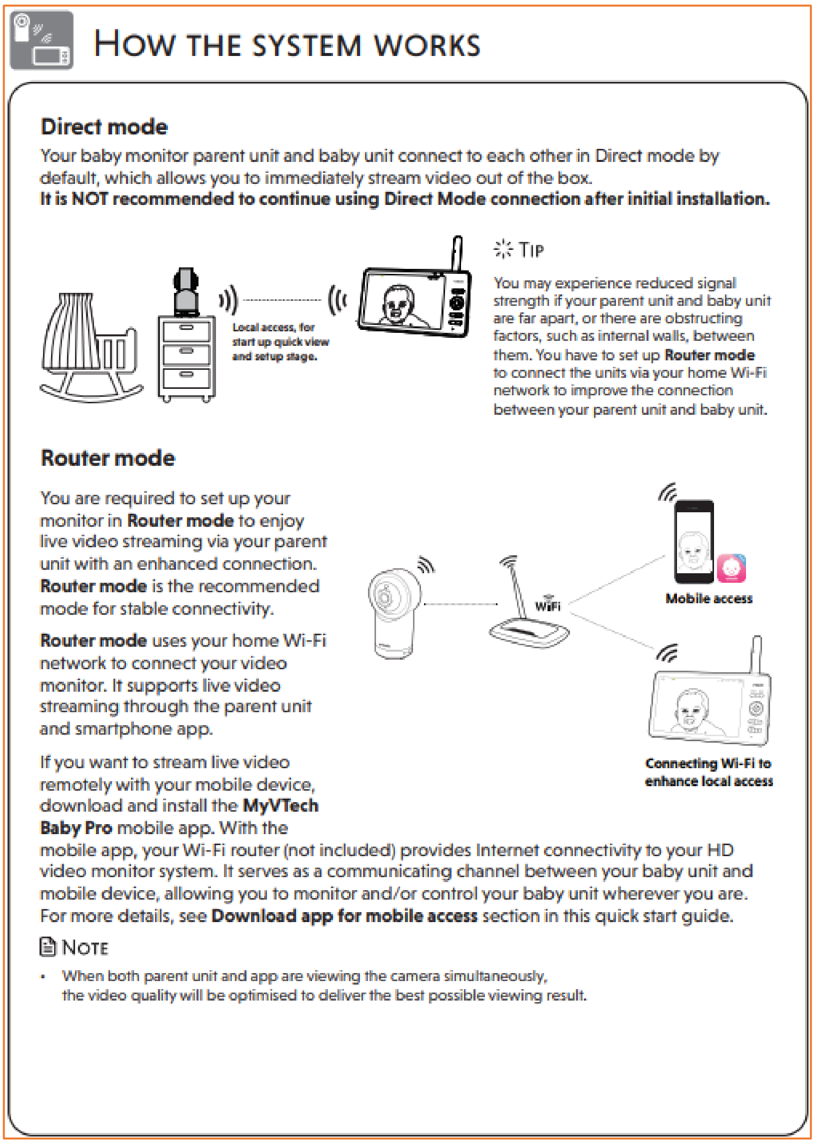


See e.g., https://play.google.com/store/apps/details?id=com.cams.VTech.mvb.pro.

See e.g., https://play.google.com/store/apps/details?id=com.cams.VTech.mvb.pro.

70. On information and belief, one or more components of the VTech Products and Services comprises a device (e.g., a baby monitoring unit) that is

connected to said Internet and that generates super-media feedback signals (e.g., video/audio signals, images, and temperature alerts).



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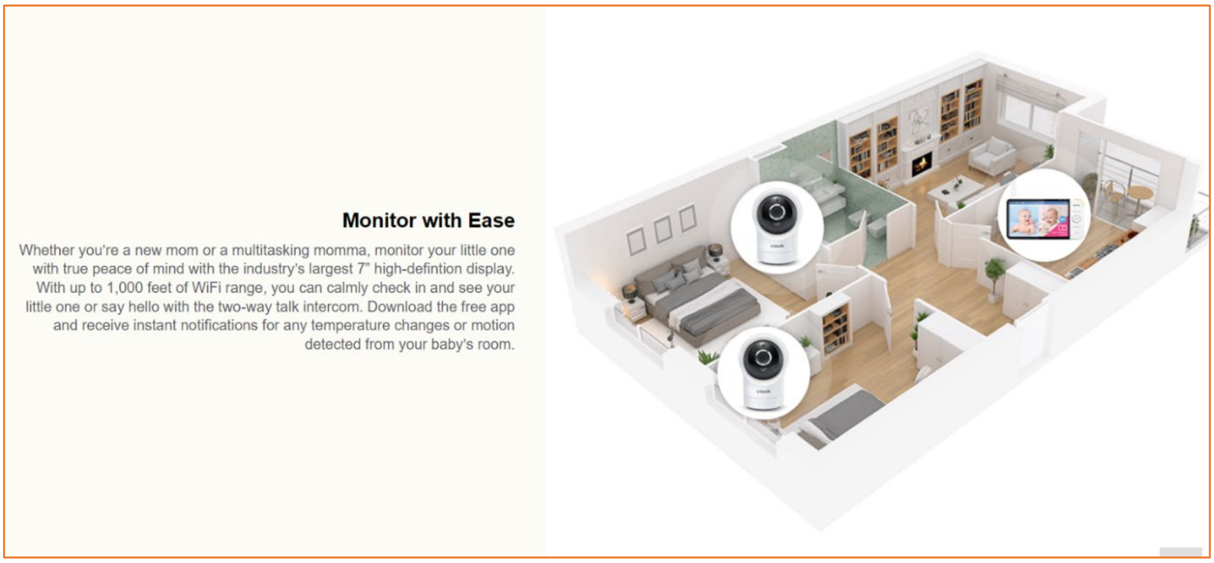
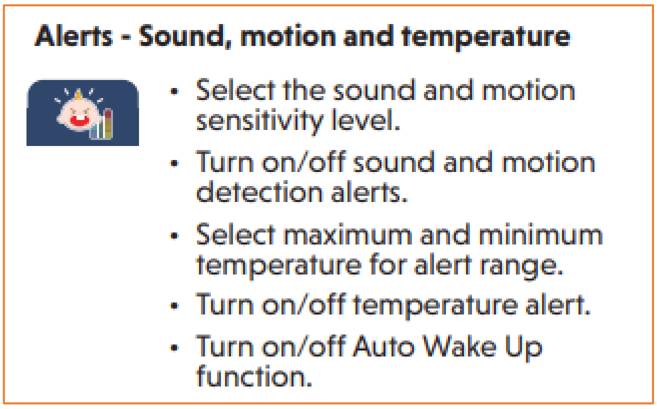


See e.g., https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.



See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

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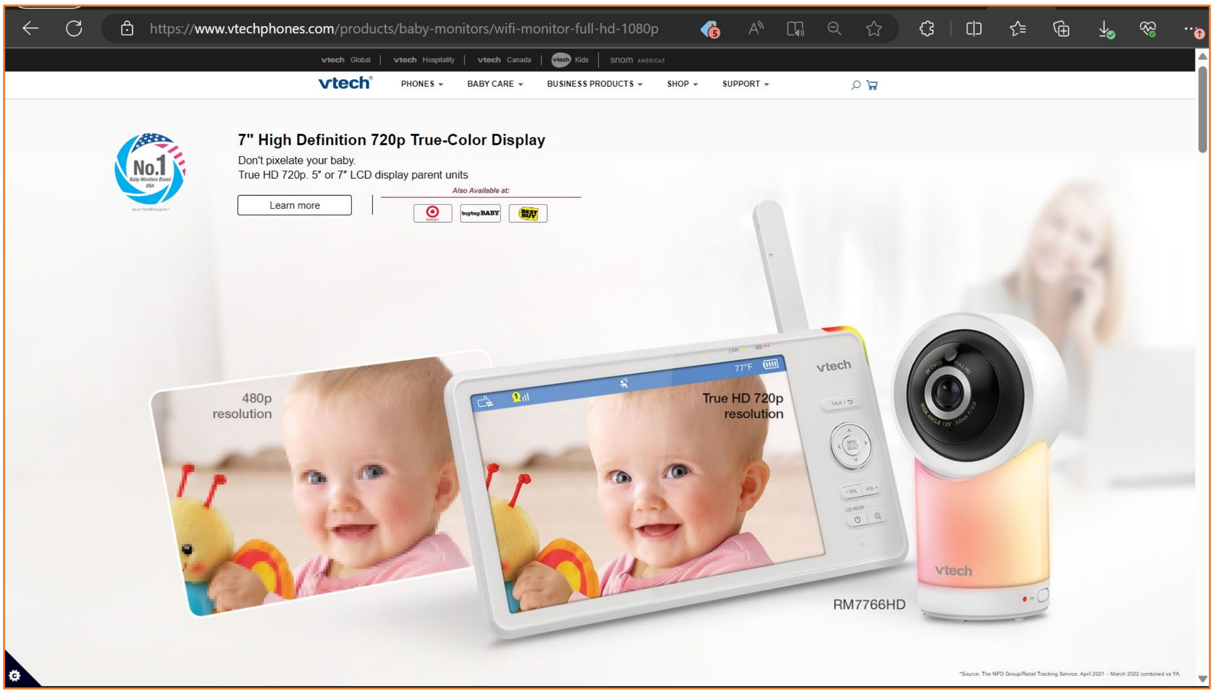
71. On information and belief, one or more components of the VTech Products and Services comprises a computer (*e.g.*, a parent unit, or a smartphone installed with 'MyVTech Baby Pro' app) connected to said Internet and a controller that provides super-media feedback, wherein said computer generates and transmits command signals (*e.g.*, alert commands) using said controller to said device and outputs said super-media feedback signals (*e.g.*, video/audio signals, images, and temperature alerts) to said controller.



See e.g., https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.

See e.g., https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.

Alerts - Sound, motion and temperature

- Select the sound and motion sensitivity level.
- Turn on/off sound and motion detection alerts.
- Select maximum and minimum temperature for alert range.
- Turn on/off temperature alert.
- Turn on/off Auto Wake Up function.

See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

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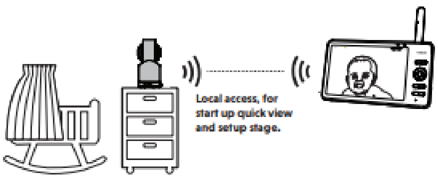

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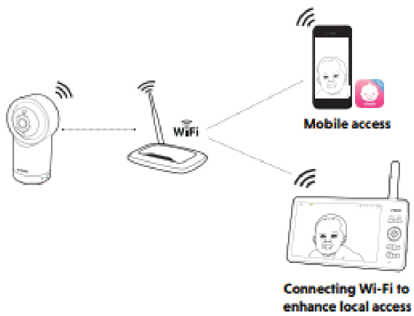

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See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

MyVTech Baby Pro

About this app

Stay in touch with your baby when you're away. With the free MyVTech Baby Pro app and your RM series baby monitor, you can watch your little one remotely—from virtually anywhere, at any time in Full HD. Maybe you want to check in on the family while traveling, or see all the fun the kids are having with the babysitter.

Enjoy continuous full HD video, 2-way conversations and motion alerts—all from up to 10 cameras.

Download the MyVTech Baby Pro app, then simply follow the in-app instructions to get started.

Use the app to:
Monitor your little one with continuous full HD video
Help calm your baby by playing melodies or say hello using 2-way talk
Control your VTech RM series pan and tilt enabled camera(s)
Receive motion alerts to let you know if the baby is up and about
Zoom the camera for 10 times more details

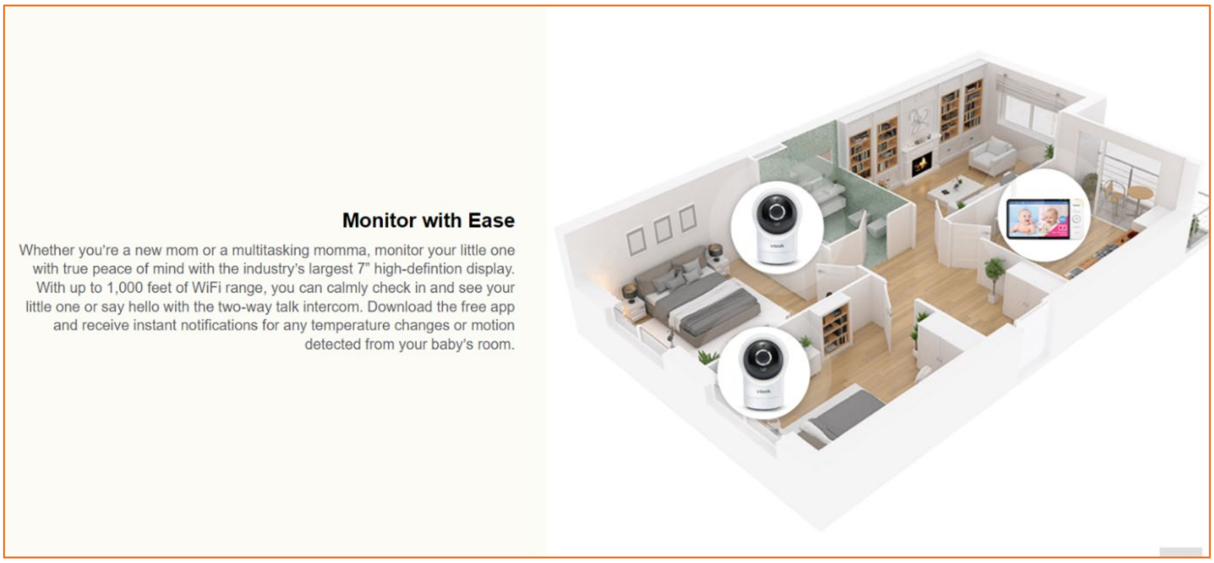
Save precious moments directly to your smart phone to share with family and friends

Please note: This app is for use with VTech RM Series Wi-Fi cameras.

See e.g., https://play.google.com/store/apps/details?id=com.cams.VTech.mvb.pro.

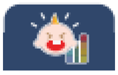
See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

72. On information and belief, one or more components of the VTech Products and Services comprises wherein said closed-loop control system is event-based (*e.g.*, each alert/recording is triggered by motion detection) to ensure stability and synchronization of said closed-loop system.



See e.g., https://www.VTechphones.com/products/baby-monitors/wifi-monitor-full-hd-1080p.

Alerts - Sound, motion and temperature



- Select the sound and motion sensitivity level.
- Turn on/off sound and motion detection alerts.
- Select maximum and minimum temperature for alert range.
- Turn on/off temperature alert.
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See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

See e.g., https://cdn-web.vtp-media.com/products/RM/RM7766/RM7766-XHD_QSG_V4_20221228.pdf.

73. ThinkLogix has been damaged by and has suffered irreparable harm as a result of Vtech's infringement.

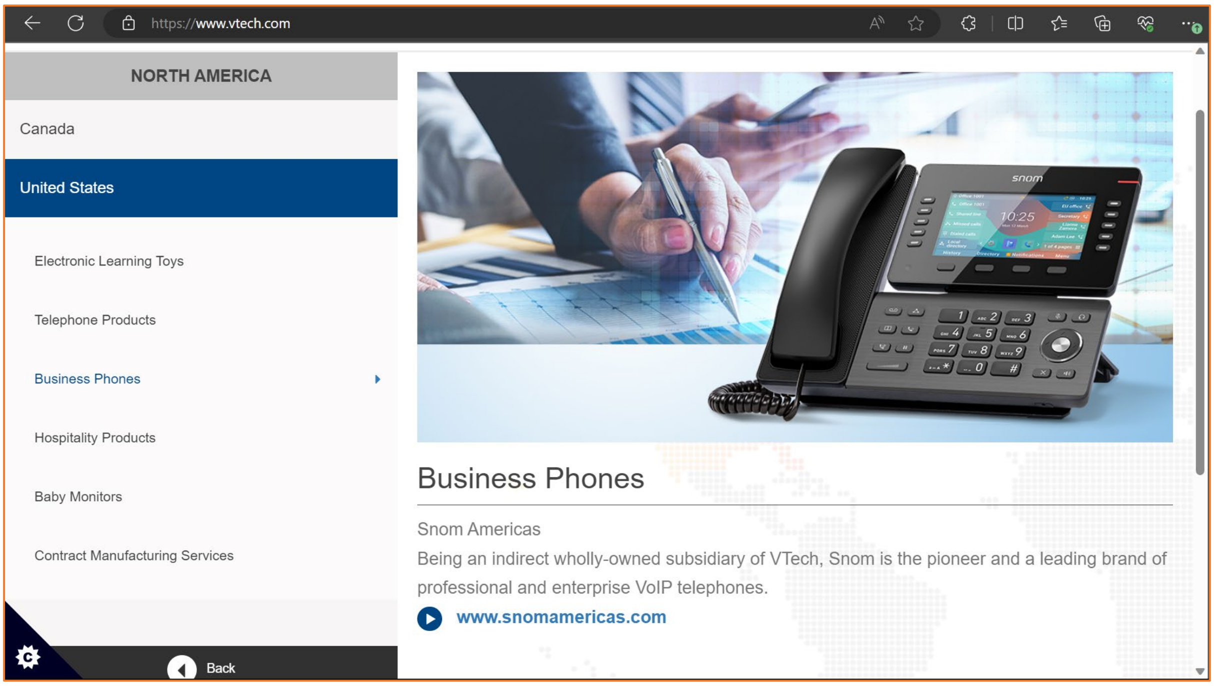
Count II – Infringement of United States Patent No. 8,599,835

74. ThinkLogix repeats, realleges, and incorporates by reference, as if fully set forth here, the allegations of the preceding paragraphs above.

75. On information and belief, VTech is in violation 35 U.S. C. § 271(a) with respect to one or more claims of the '835 patent.

76. On information and belief, VTech (or those acting on its behalf) has made, used, offered to sell, sold, and/or imported one or more components of the VTech Products and Services that infringes (literally and/or under the doctrine of equivalents) at least claim 1 of the '835 patent.

77. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method comprising the step of receiving a first SIP message (*e.g.*, a SIP INVITE), from a client (*e.g.*, a user initiating or attending the SIP-based call via VTech's '2-Line Color SIP Cordless Accessory Handset, or VTech Snom's Desk Telephone D717, or similar VTech's desk phones), requesting media (*e.g.*, requesting for a live SIP-based audio call).



See e.g., https://www.VTech.com.

D785 Desk Telephone
True paperless

- High-resolution color display
- Second screen as graphical display
- 6 configurable, self-labeling keys
- Gigabit switch | USB port
- 12 SIP identities | Sensor hook switch
- Bluetooth-compatible
- Dual Stack IPv4/IPv6

D765 Desk Telephone

- High-resolution color display
- 16 multicolor function keys
- Gigabit switch | USB port
- 12 SIP identities | Sensor hook switch
- Bluetooth-compatible
- Dual Stack IPv4/IPv6

D735 Desk Telephone

- High-resolution color display
- 8 configurable, self-labeling keys
- Motion sensor
- USB port
- Wideband audio
- Dual Stack IPv4/IPv6

D717 Desk Telephone

- High-resolution color display
- 3 configurable, self-labeling keys
- USB port
- Speakerphone
- Wideband audio
- Dual Stack IPv4/IPv6

D715 Desk Telephone

- Graphical display with backlight
- Gigabit switch | USB port
- 4 SIP identities | Sensor hook switch
- Wideband audio
- Dual Stack IPv4/IPv6

D120 Desk Telephone

- Graphical display with backlight
- Low power consumption (PoE)
- 2 SIP identities | Sensor hook switch
- Wideband audio
- Dual Stack IPv4/IPv6

D7 black (Expansion module)

- High-resolution display with backlight
- 18 configurable, self-labeling LED keys
- Power supplied by phone via USB port
- Daisy-chain up to 3 expansion modules

See e.g., https://www.snomamericas.com/assets/1e3c57af-0e3e-46f0-a488-d857429943ad/snomamericas_product_catalog_en.pdf

Global 700 Series Blackline

	D785	D765	D735	D717	D715	D120
Display	Color TFT	Color TFT	Color TFT	Color TFT	Graphical	Graphical
SIP Identities	12	12	12	6	4	2
Ethernet/Switch	2 x Gigabit	2 x Gigabit	2 x Gigabit	2x Gigabit	2 x Gigabit	2x 10/100 Mbps
USB	●	●	●	●	●	○
Programmable Keys (LED)	6 multicolor	16 multicolor	8 multicolor	3 multicolor	5	2
Expansion Module	●	●	●	●	●	○
Position	Dual-angle 46° and 28°	Dual-angle 46° and 28°	Dual-angle 46° and 28°	Dual-angle 46° and 28°	Dual-angle 46° and 28°	Single-angle 30°
Color	Black	Black	Black	Black	Black	Black
Bluetooth-compatible	Integrated	Integrated	○	○	Via USB dongle	○

See e.g., https://www.snomamericas.com/assets/1e3c57af-0e3e-46f0-a488-d857429943ad/snomamericas_product_catalog_en.pdf.

See e.g., https://www.snomamericas.com/en/pd/ip-phones/desk-phones/d7xx-series-next-gen/d717.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)(2)
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold(2)
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup(2)
- Call park, call unpark(2)
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)(2)
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

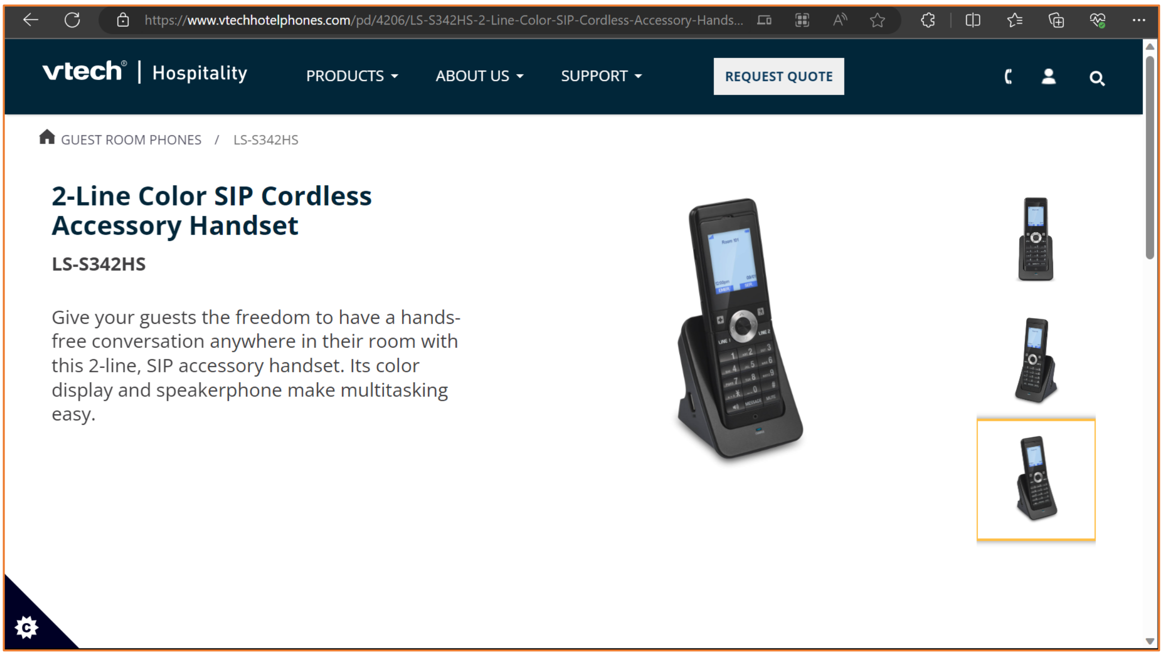
User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
- Password-protected web interface
- VPN support
- VLAN (IEEE 802.1Q)
- LLDP-MED, RTCP-XR

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.



See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

https://cdn-web.vtp-media.com/products/LS/LS-S3410-USB/LS-S34X0_UG_V6_20210223.pdf

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

The cordless handset and the telephone base cannot be used on the same call. However, calls can be switched between the cordless handset and the telephone base speakerphone.

When the cordless handset or the telephone base is in use, the **TALK** key on the cordless handset and the **LINE** key on the telephone base illuminate.

Receive a call

When there is an incoming call, the telephone rings. The **LINE** key and the **MESSAGE WAITING** LED on the telephone base flash, and the **TALK** key on the cordless handset flashes.

To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or 🔊**/SPEAKER.** The **TALK** key illuminates when phone line is in use. The 🔊**/SPEAKER** key illuminates when in speakerphone mode.

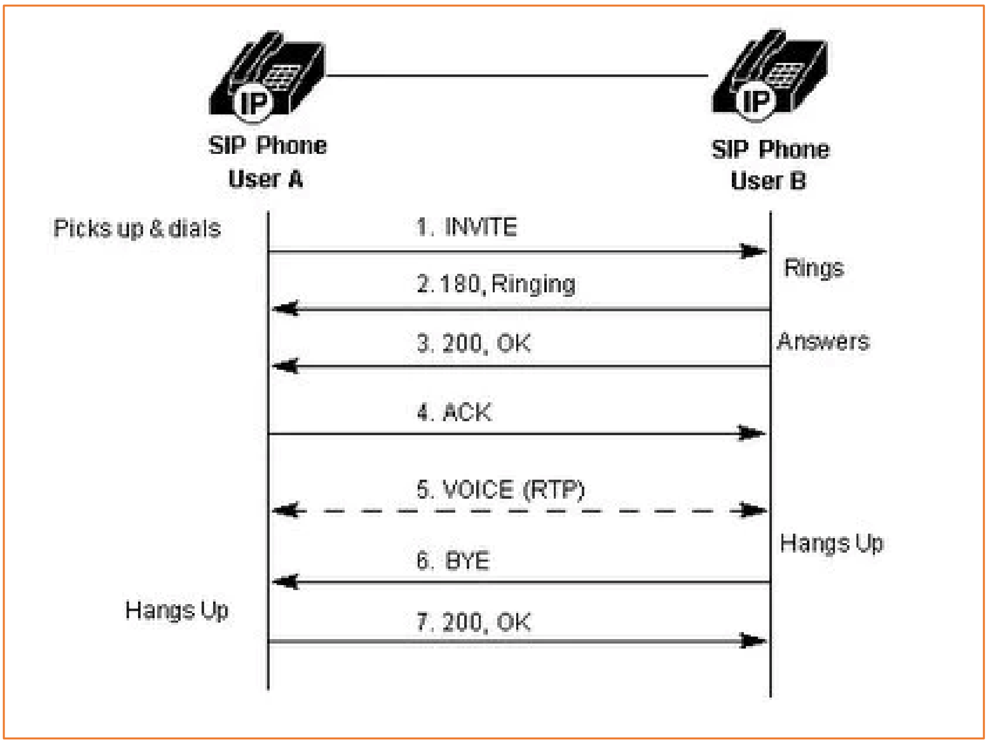
To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The 🔊**/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or 🔊**/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The 🔊**/SPEAKER** key illuminates when in speakerphone mode.

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.



See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

78. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method comprising the step of sending, using a processor, a notify SIP message (*e.g.*, 200 OK) to the client, wherein the notify SIP message comprises a contact header, and wherein the contact header comprises a parameter (*e.g.*, Session ID) that indicates the requested media is to be streamed to the client.

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

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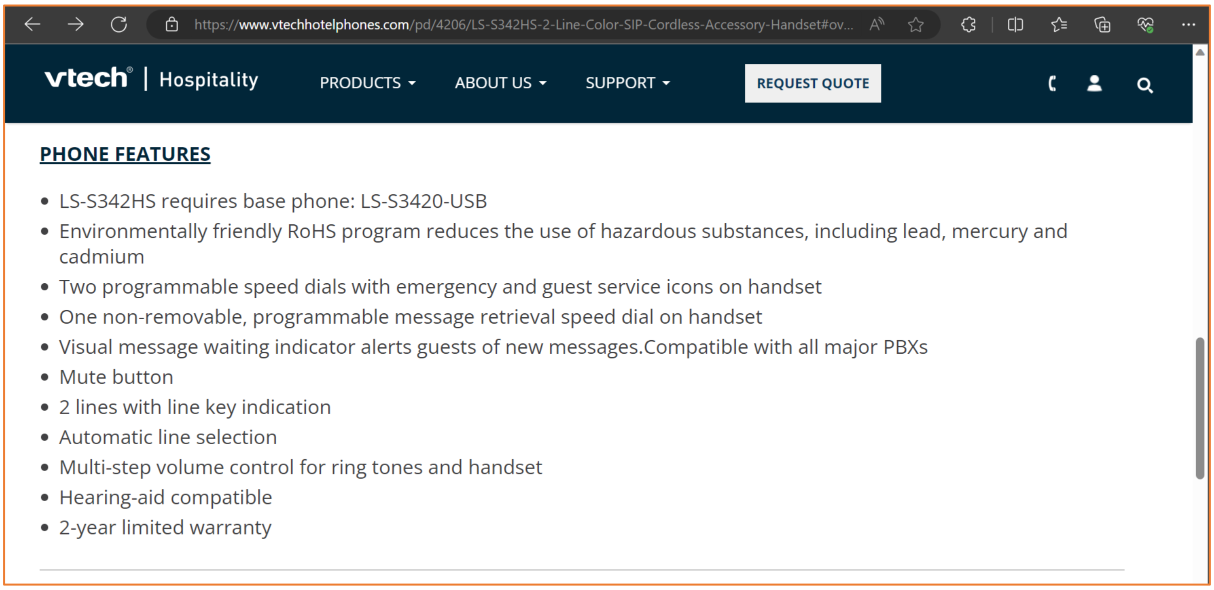


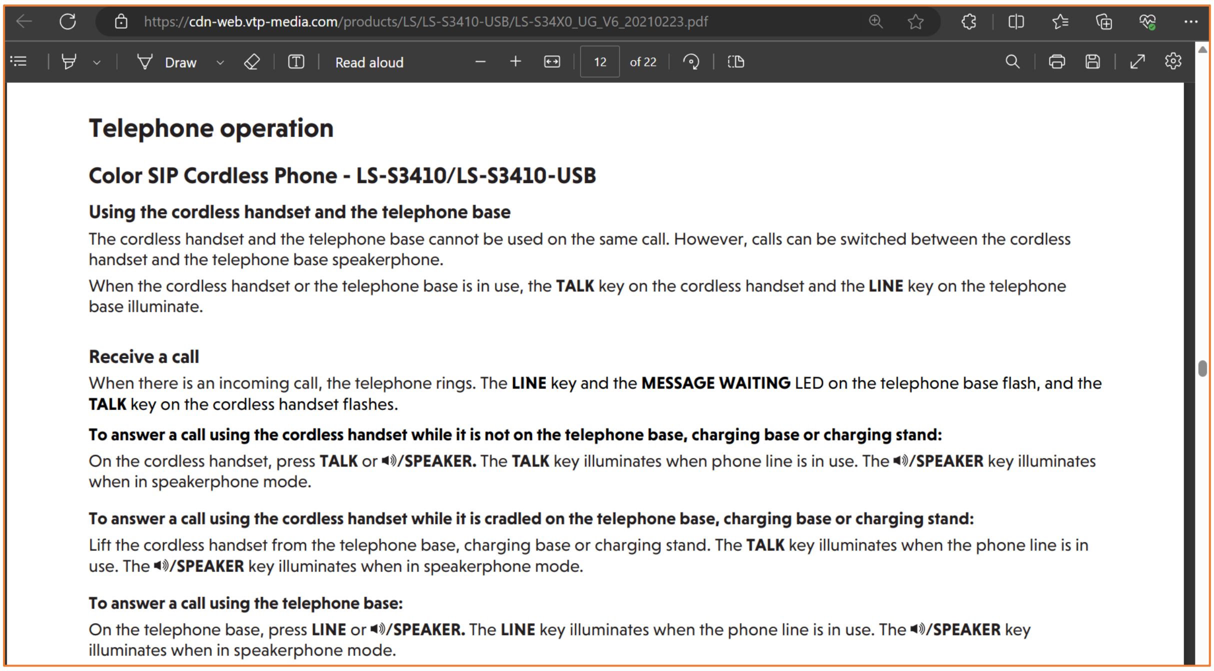
https://www.vtechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#ov...

vtech | Hospitality PRODUCTS ABOUT US SUPPORT REQUEST QUOTE

PHONE FEATURES

- LS-S342HS requires base phone: LS-S3420-USB
- Environmentally friendly RoHS program reduces the use of hazardous substances, including lead, mercury and cadmium
- Two programmable speed dials with emergency and guest service icons on handset
- One non-removable, programmable message retrieval speed dial on handset
- Visual message waiting indicator alerts guests of new messages.Compatible with all major PBXs
- Mute button
- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
- 2-year limited warranty

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.



https://cdn-web.vtp-media.com/products/LS/LS-S3410-USB/LS-S34X0_UG_V6_20210223.pdf

Draw Read aloud 12 of 22

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

The cordless handset and the telephone base cannot be used on the same call. However, calls can be switched between the cordless handset and the telephone base speakerphone.

When the cordless handset or the telephone base is in use, the **TALK** key on the cordless handset and the **LINE** key on the telephone base illuminate.

Receive a call

When there is an incoming call, the telephone rings. The **LINE** key and the **MESSAGE WAITING** LED on the telephone base flash, and the **TALK** key on the cordless handset flashes.

To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or **/SPEAKER.** The **TALK** key illuminates when phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

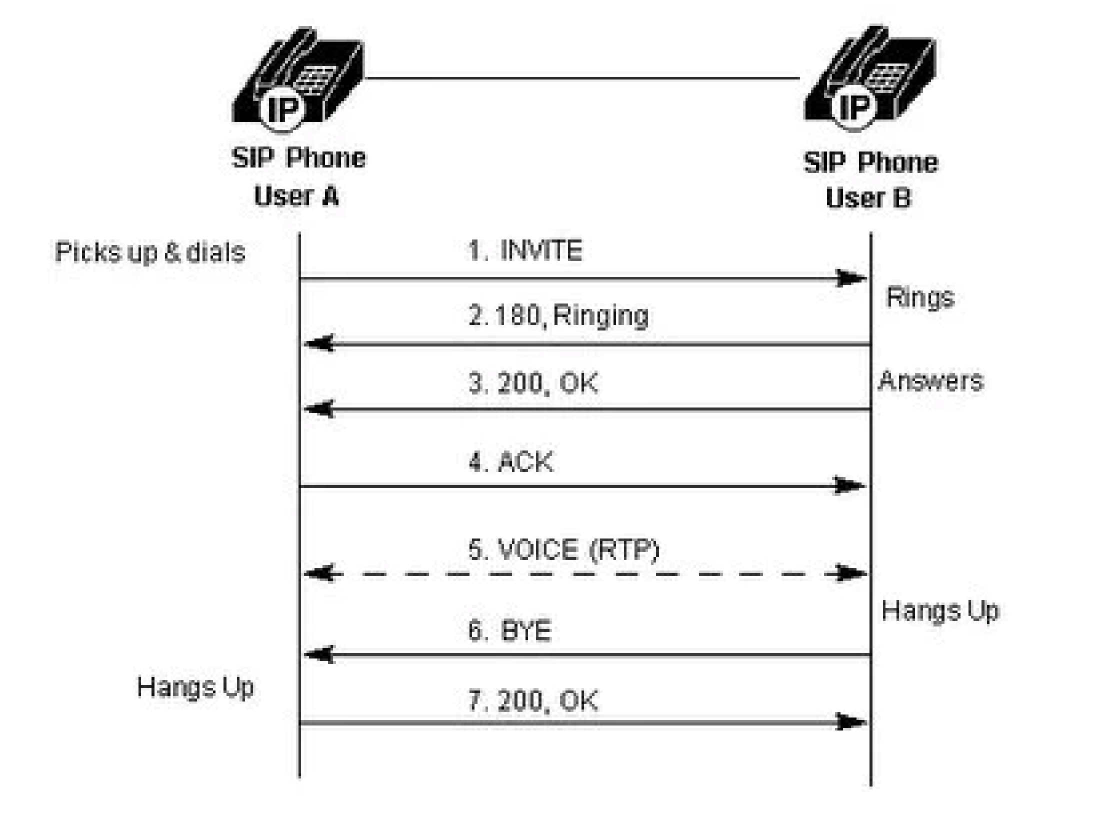
To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or **/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.



See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

79. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method comprising the step of streaming the requested media to the client based in part on the parameter.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)[(2)]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[(2)]
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[(2)]
- Call park, call unpark[(2)]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[(2)]
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
- Password-protected web interface
- VPN support
- VLAN (IEEE 802.1Q)
- LLDP-MED, RTCP-XR

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https://cdn-web.vtp-media.com/products/LS/LS-S3410-USB/LS-S34X0_UG_V6_20210223.pdf

Read aloud 12 of 22

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

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To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or **/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

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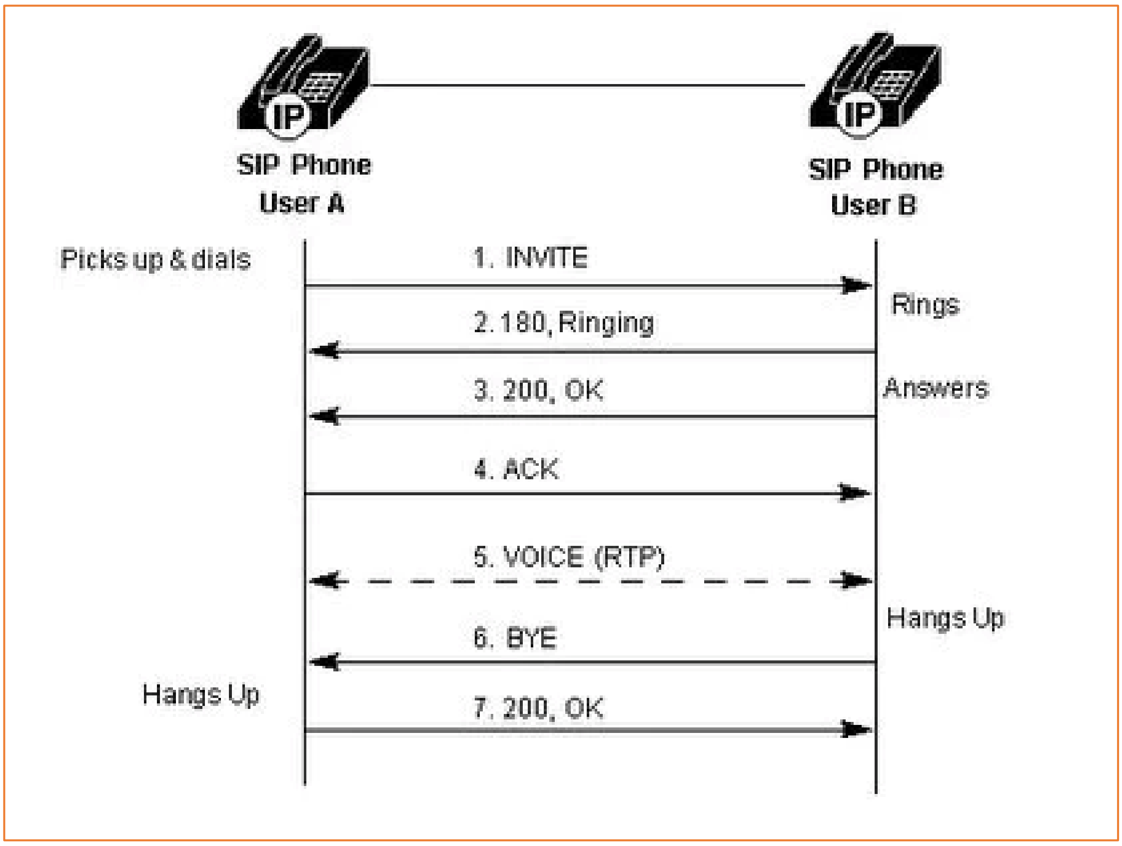

See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

80. On information and belief, VTech is in violation of 35 U.S.C. § 271(b) and has been, at least since its October 2023 knowledge of the ’835 patent, indirectly infringing and continues to indirectly infringe at least claim 1 of the ’835 patent by knowingly and specifically intending to induce infringement by others (including, without limitation, VTech’s customers) and possessing specific intent to encourage infringement by VTech’s users of VTech’s ‘2-Line Color SIP Cordless Accessory Handset, or VTech Snom’s Desk Telephone D717, or similar VTech’s desk phones. The components of the VTech System, are specifically configured to function in accordance with the ’835 patent claims and are material parts of the invention.

81. ThinkLogix has been damaged by and has suffered irreparable harm as a result of Vtech’s infringement.

Count III – Infringement of United States Patent No. 9,231,994

82. ThinkLogix repeats, realleges, and incorporates by reference, as if fully set forth here, the allegations of the preceding paragraphs above.

83. On information and belief, VTech is in violation 35 U.S. C. § 271(a) with respect to one or more claims of the ’994 patent.

84. On information and belief, VTech (or those acting on its behalf) has made, used, offered to sell, sold, and/or imported one or more components of the VTech Products and Services that infringes (literally and/or under the doctrine of equivalents) at least claim 19 of the ’994 patent.

85. On information and belief, one or more components of the VTech Products and Services comprises one or more processors configured to send a first session invitation protocol (SIP) message (*e.g.*, SIP INVITE) from a first device (*e.g.*, 2-Line Color SIP Cordless Accessory Handset and VTech Snom's Desk Telephone D717 and other VTech devices) to a second device (*e.g.*, VTech's handset or another VTech's device receiving the call), wherein a first header field of the first SIP message includes a first indicator indicating a request for a media stream (*e.g.*, initiating, or attending a SIP-based audio call) and a source (*e.g.*, Request-URI) of the requested media stream, wherein the first SIP message comprises a contact header and wherein the contact header comprises a parameter (*e.g.*, Content identifier, or PSI) that indicates the requested media is to be streamed to the first device.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)[2]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[2]
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[2]
- Call park, call unpark[2]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[2]
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
- Password-protected web interface
- VPN support
- VLAN (IEEE 802.1Q)
- LLDP-MED, RTCP-XR

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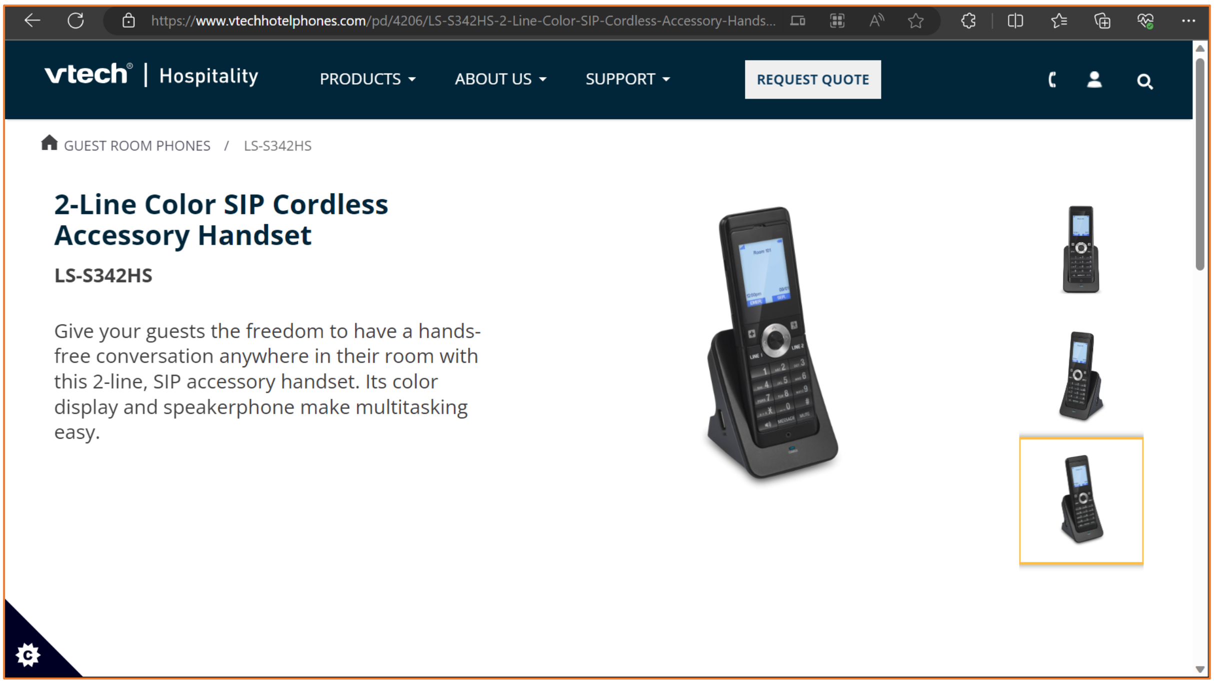

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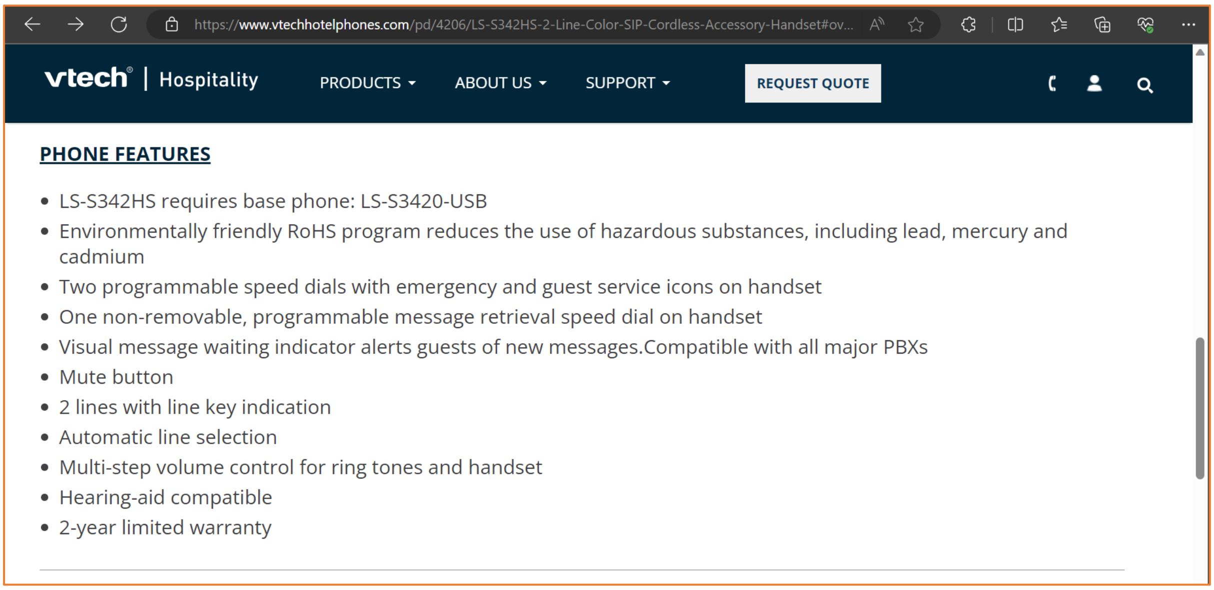

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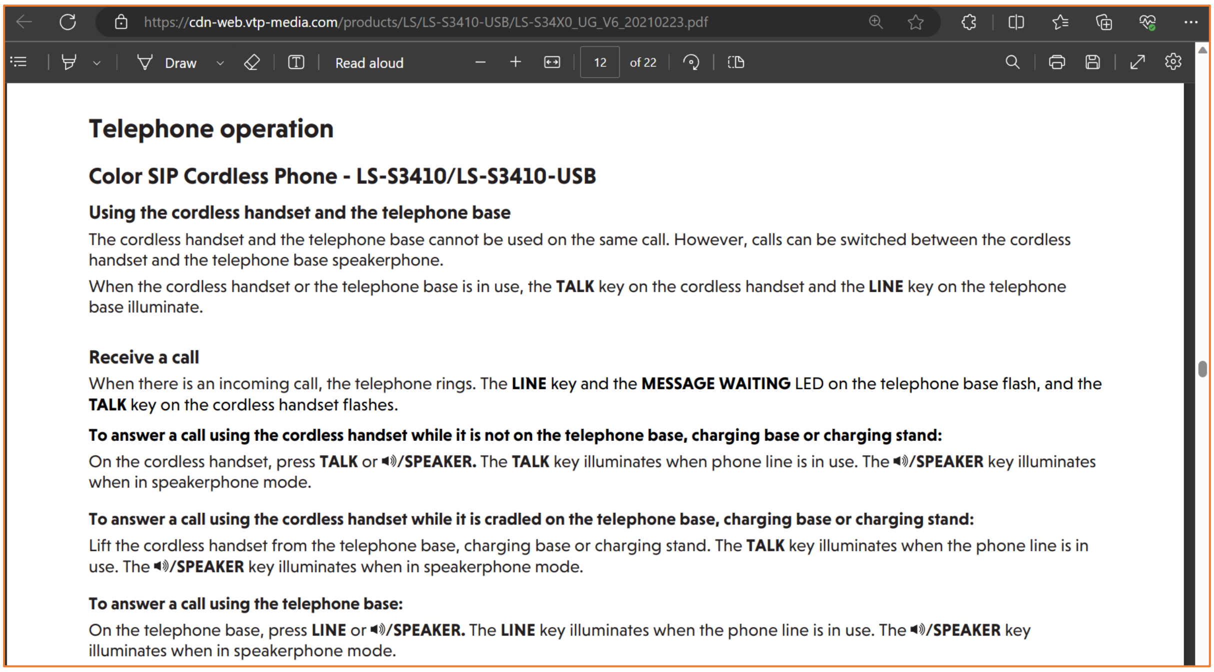

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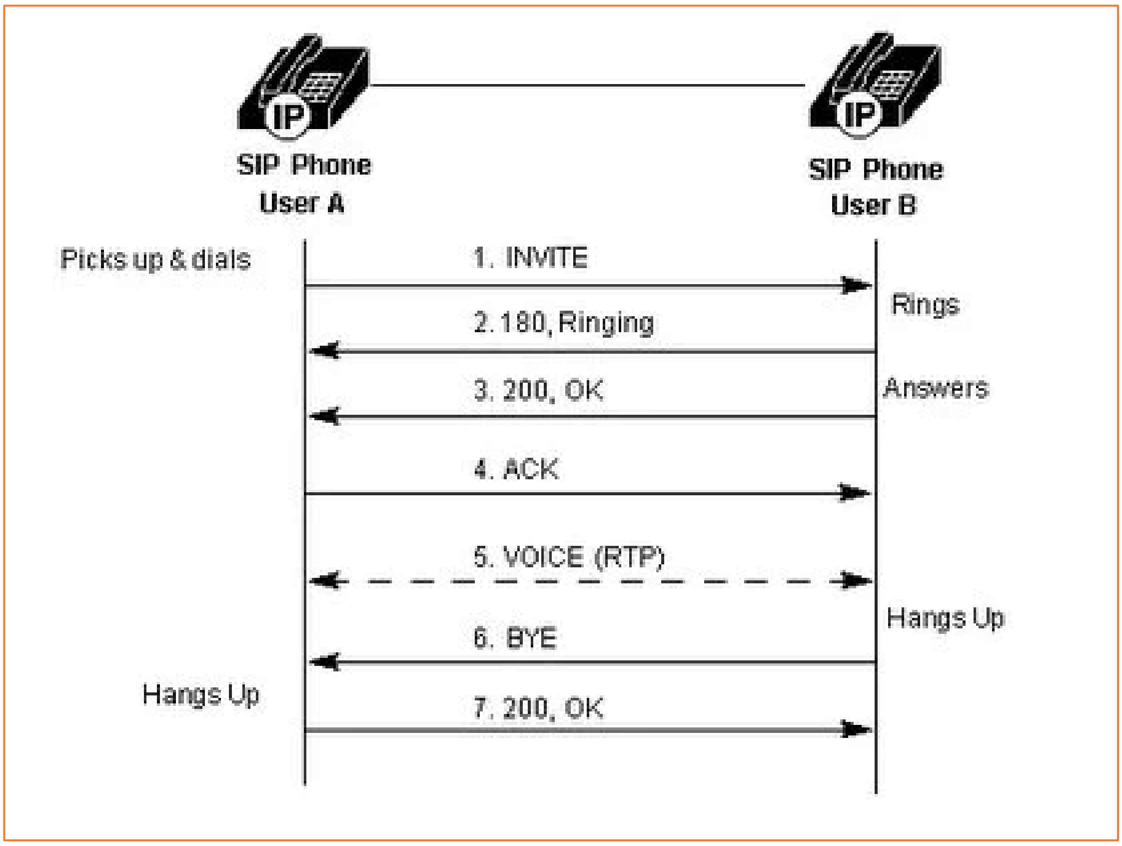

See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

86. On information and belief, one or more components of the VTech Products and Services comprises one or more processors configured to receive a second SIP message (*e.g.*, 200 OK) from the second device at the first device, wherein a second header field of the second SIP message includes a second indicator indicating acceptance of the request for the media stream.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)[2]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[2]
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[2]
- Call park, call unpark[2]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[2]
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
- Password-protected web interface
- VPN support
- VLAN (IEEE 802.1Q)
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https://www.vtechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#ov...

vtech® | Hospitality PRODUCTS ABOUT US SUPPORT REQUEST QUOTE

PHONE FEATURES

- LS-S342HS requires base phone: LS-S3420-USB
- Environmentally friendly RoHS program reduces the use of hazardous substances, including lead, mercury and cadmium
- Two programmable speed dials with emergency and guest service icons on handset
- One non-removable, programmable message retrieval speed dial on handset
- Visual message waiting indicator alerts guests of new messages.Compatible with all major PBXs
- Mute button
- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
- 2-year limited warranty

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

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Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

The cordless handset and the telephone base cannot be used on the same call. However, calls can be switched between the cordless handset and the telephone base speakerphone.

When the cordless handset or the telephone base is in use, the **TALK** key on the cordless handset and the **LINE** key on the telephone base illuminate.

Receive a call

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To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or **/SPEAKER.** The **TALK** key illuminates when phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

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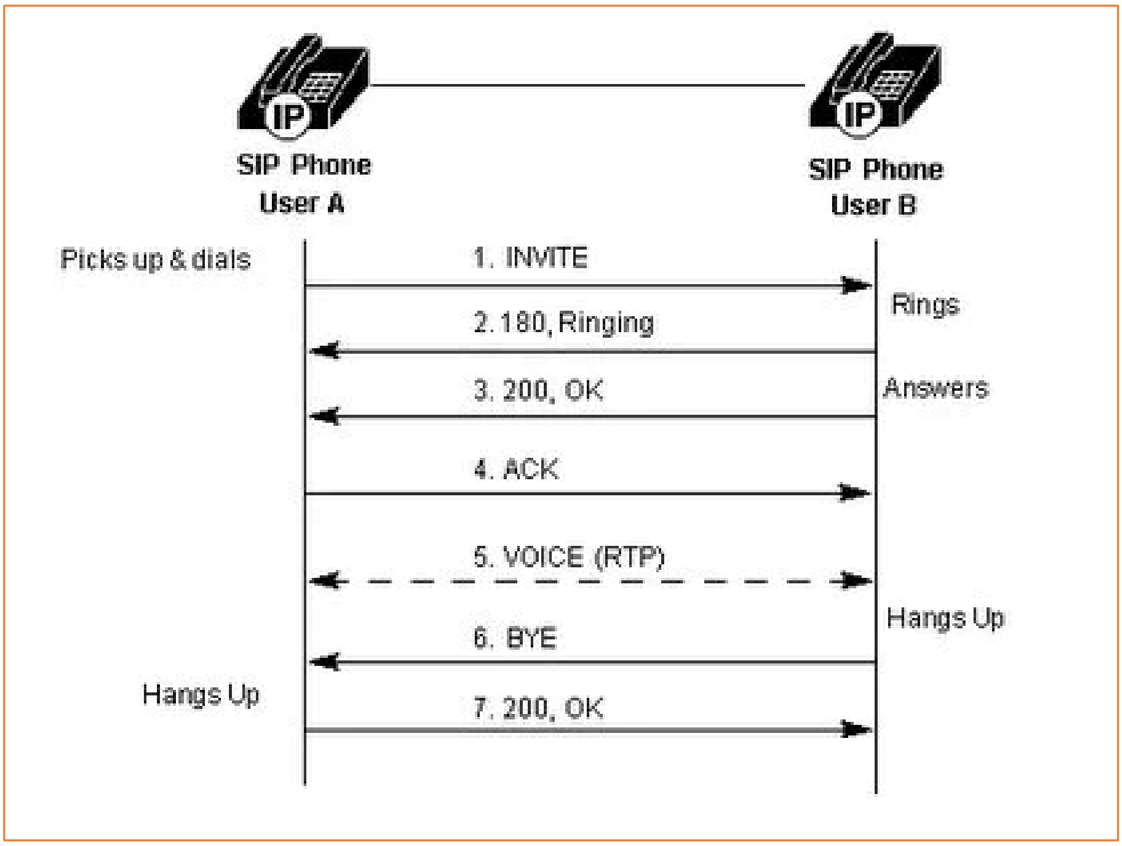

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87. On information and belief, one or more components of the VTech Products and Services comprises one or more processors configured to send a third SIP (*e.g.,* ACK message) message from the first device to the second device, wherein a third header field of the third SIP message includes a third indicator indicating that the second device initiate streaming of the requested media stream.

snom

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D717

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Audio

- Codecs:

Specifications

Protocols

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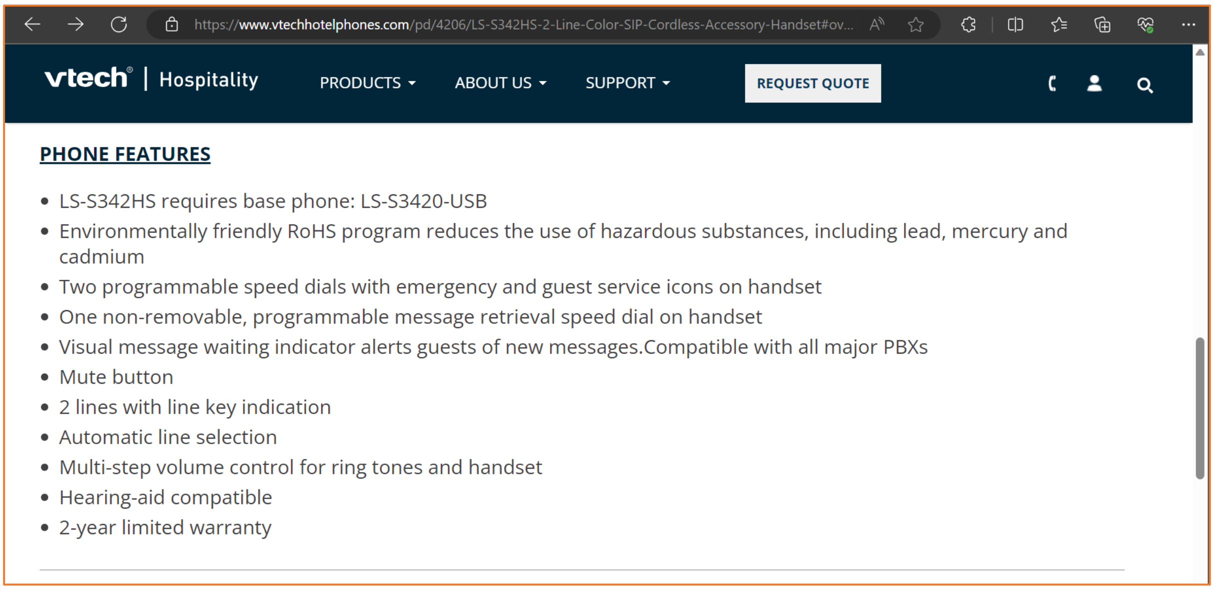

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- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
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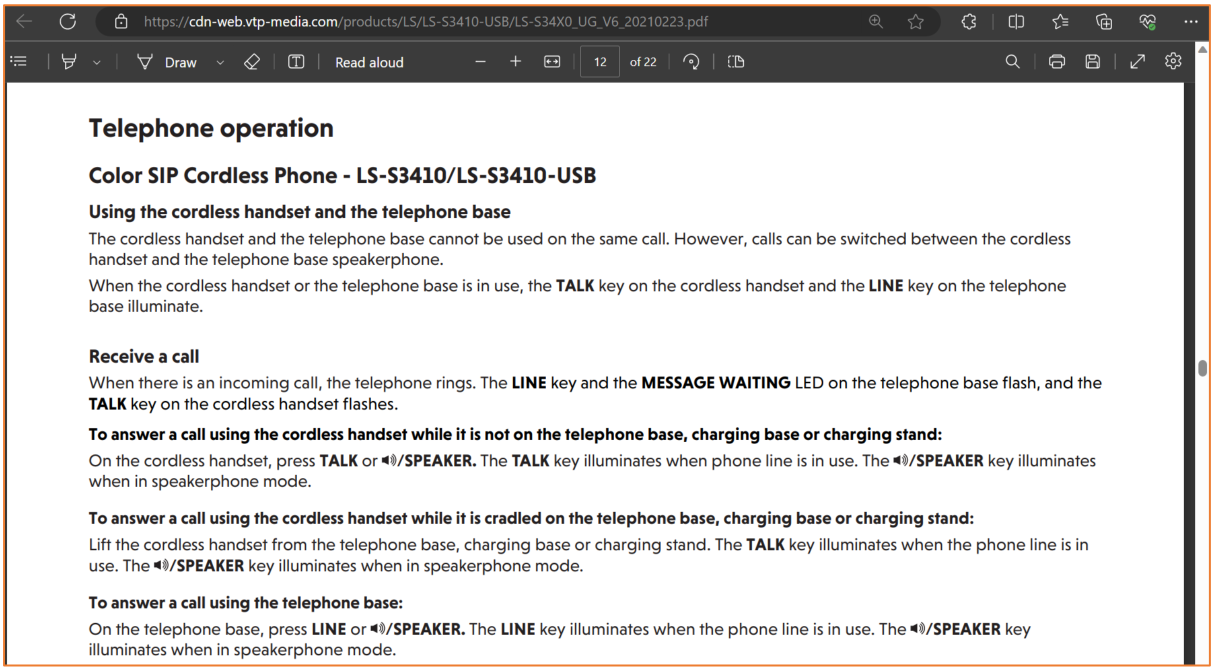

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Draw Read aloud 12 of 22

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

The cordless handset and the telephone base cannot be used on the same call. However, calls can be switched between the cordless handset and the telephone base speakerphone.

When the cordless handset or the telephone base is in use, the **TALK** key on the cordless handset and the **LINE** key on the telephone base illuminate.

Receive a call

When there is an incoming call, the telephone rings. The **LINE** key and the **MESSAGE WAITING** LED on the telephone base flash, and the **TALK** key on the cordless handset flashes.

To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or **/SPEAKER.** The **TALK** key illuminates when phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

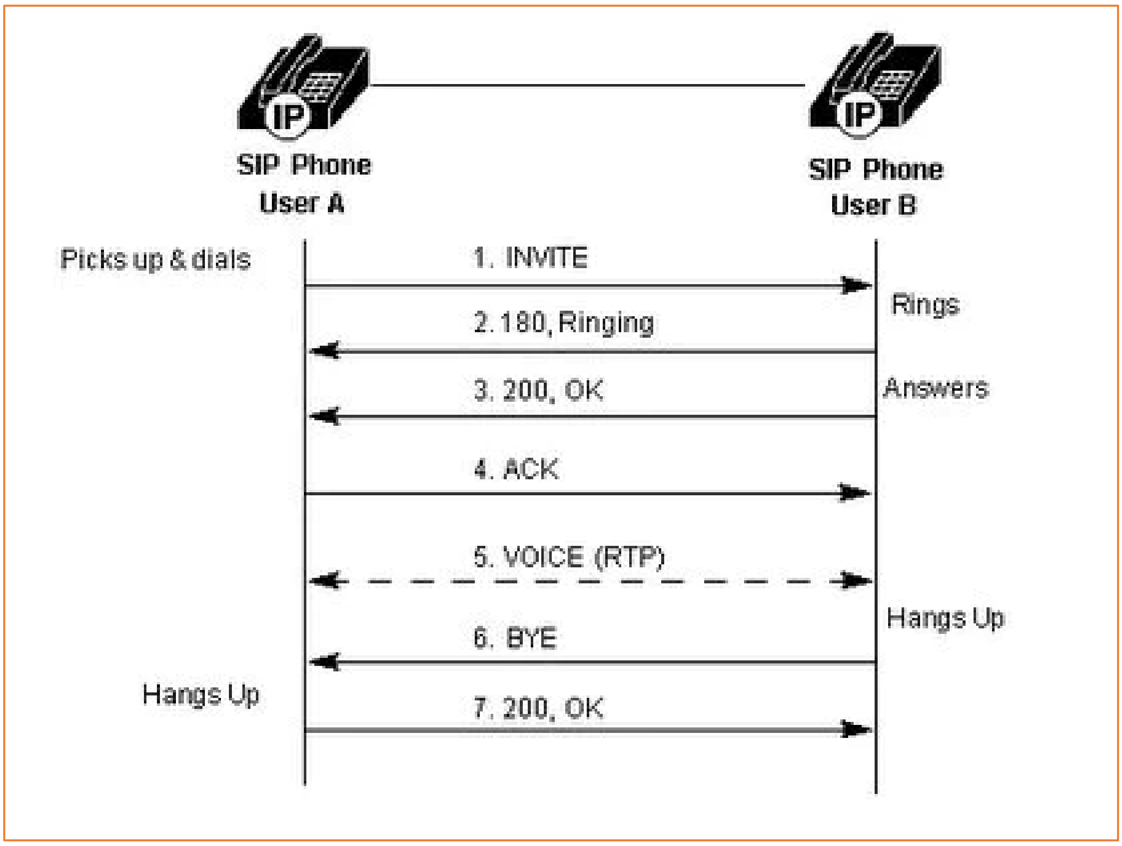
To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or **/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.



See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

88. On information and belief, one or more components of the VTech Products and Services comprise one or more processors configured to receive the requested media stream (*e.g.*, audio etc.) from the second device at the first device.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)[2]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[2]
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[2]
- Call park, call unpark[2]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[2]
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
- Password-protected web interface
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https://www.vtechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#ov...

vtech® | Hospitality PRODUCTS ABOUT US SUPPORT REQUEST QUOTE

PHONE FEATURES

- LS-S342HS requires base phone: LS-S3420-USB
- Environmentally friendly RoHS program reduces the use of hazardous substances, including lead, mercury and cadmium
- Two programmable speed dials with emergency and guest service icons on handset
- One non-removable, programmable message retrieval speed dial on handset
- Visual message waiting indicator alerts guests of new messages.Compatible with all major PBXs
- Mute button
- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
- 2-year limited warranty

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https://cdn-web.vtp-media.com/products/LS/LS-S3410-USB/LS-S34X0_UG_V6_20210223.pdf

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Receive a call

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To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or **/SPEAKER.** The **TALK** key illuminates when phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or **/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

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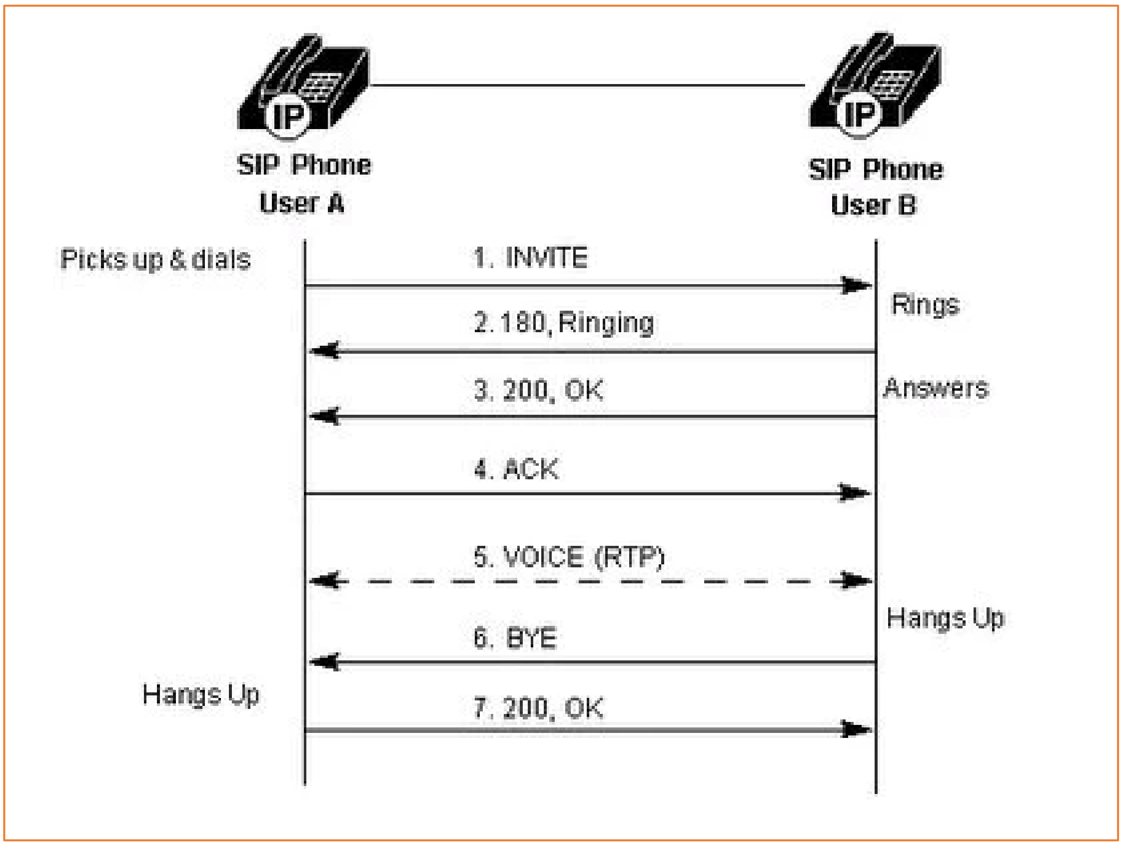

See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

89. ThinkLogix has been damaged by and has suffered irreparable harm as a result of Vtech's infringement.

<u>Count IV – Infringement of United States Patent No. 9,906,573</u>

90. ThinkLogix repeats, realleges, and incorporates by reference, as if fully set forth here, the allegations of the preceding paragraphs above.

91. On information and belief, VTech is in violation 35 U.S. C. § 271(a) with respect to one or more claims of the '573 patent.

92. On information and belief, VTech (or those acting on its behalf) has made, used, offered to sell, sold, and/or imported one or more components of the

VTech Products and Services that infringes (literally and/or under the doctrine of equivalents) at least claim 18 of the ’573 patent.

93. On information and belief, one or more components of the VTech Products and Services comprises one or more processors configured to send a first session invitation protocol (SIP) message (*e.g.*, SIP INVITE) to a media source device (*e.g.*, VTech’s 2-Line Color SIP Cordless Accessory Handset and VTech Snom’s Desk Telephone D717 receiving SIP-based audio call), wherein a first header field of the first SIP message includes a first indicator indicating a request for a media stream (*e.g.*, initiating, or attending SIP-based audio call), wherein the first indicator includes a uniform resource identifier (URI) identifying (*e.g.*, Request-URI) a source of the requested media stream and wherein a parameter (*e.g.*, Content identifier, or PSI) of the URI indicates that streaming is to be used.

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High-resolution color display and self-labeling keys

D717

Key Features

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- Wideband audio
- USB port
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Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)(2)
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold(2)
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup(2)
- Call park, call unpark(2)
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)(2)
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
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- Password-protected web interface
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https://www.vtechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#ov...

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PHONE FEATURES

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- One non-removable, programmable message retrieval speed dial on handset
- Visual message waiting indicator alerts guests of new messages.Compatible with all major PBXs
- Mute button
- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
- 2-year limited warranty

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Draw Read aloud 12 of 22

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

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Receive a call

When there is an incoming call, the telephone rings. The **LINE** key and the **MESSAGE WAITING** LED on the telephone base flash, and the **TALK** key on the cordless handset flashes.

To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or 🔈**/SPEAKER.** The **TALK** key illuminates when phone line is in use. The 🔈**/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The 🔈**/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or 🔈**/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The 🔈**/SPEAKER** key illuminates when in speakerphone mode.

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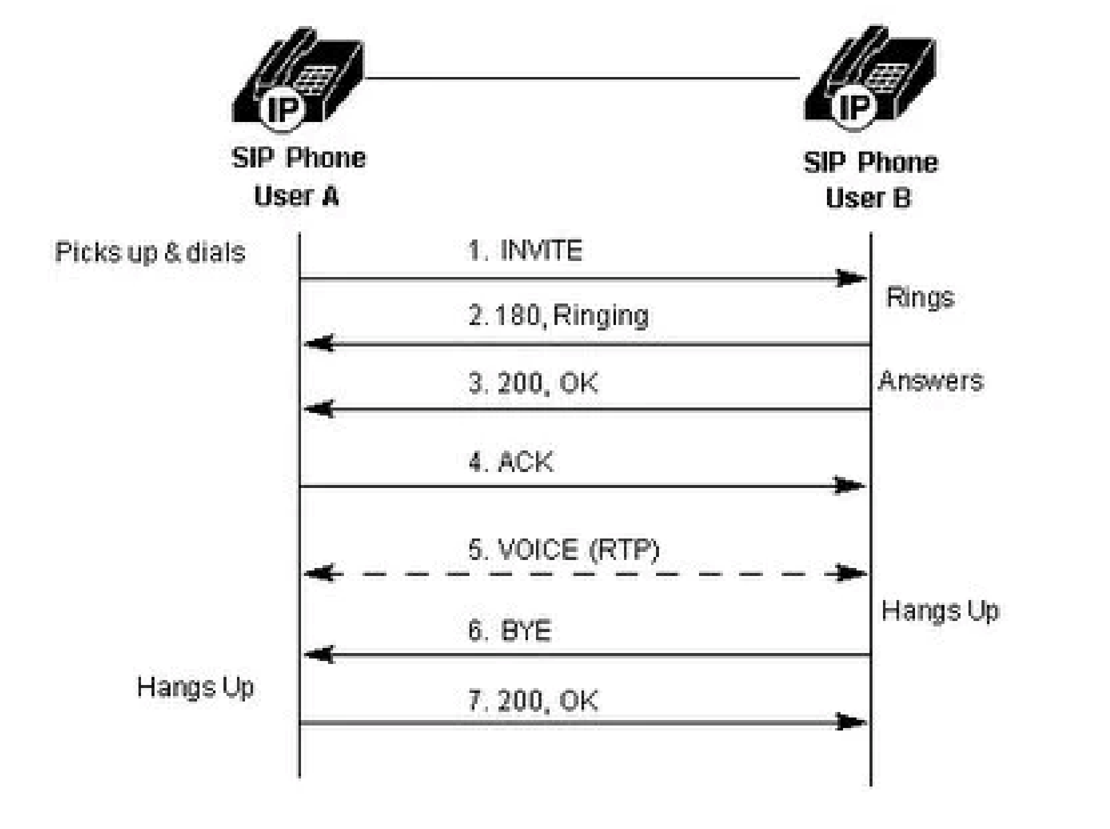

See e.g., https://www.onsip.com/voip-resources/voip-solutions/sip-call-flow-explained.

94. On information and belief, one or more components of the VTech Products and Services comprises one or more processors configured to send a second SIP message (*e.g.,* ACK message) to the media source device, wherein a header field of the second SIP message includes a second indicator indicating that the media source device initiate streaming of the requested media stream.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

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Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)[2]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[2]
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[2]
- Call park, call unpark[2]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[2]
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
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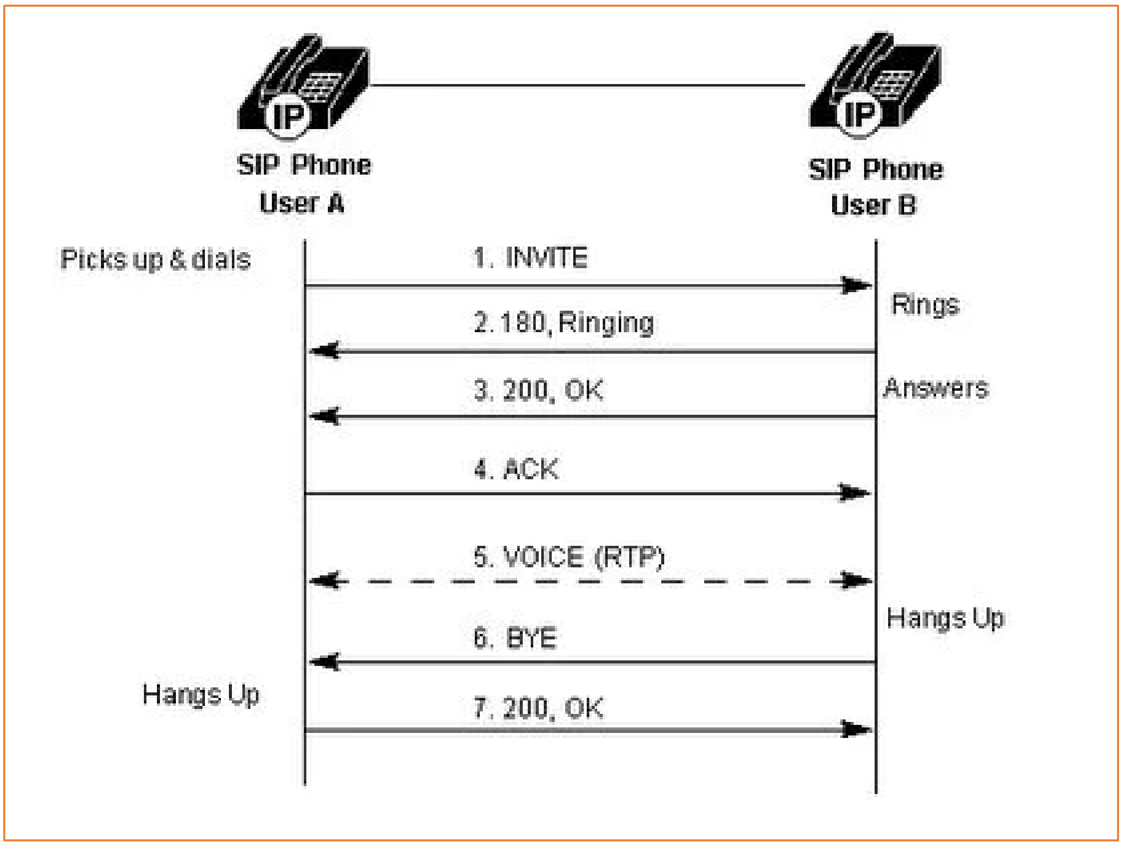

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95. On information and belief, one or more components of the VTech System comprises one or more processors configured to receive the requested media stream from the media source device.

snom

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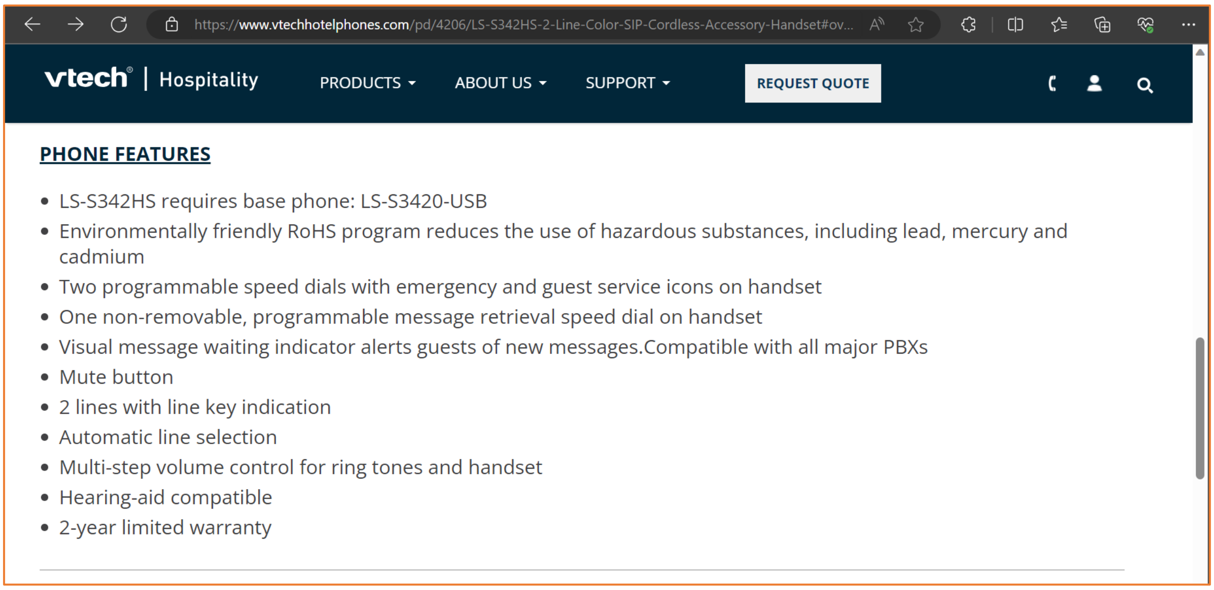

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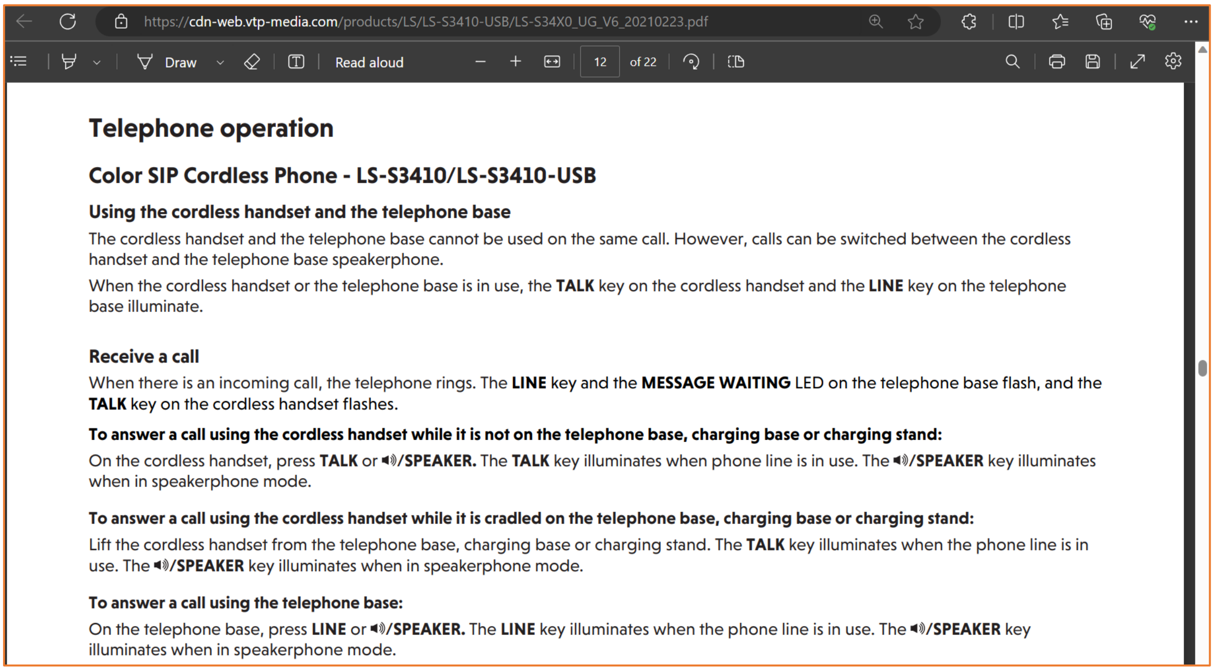
- LS-S342HS requires base phone: LS-S3420-USB
- Environmentally friendly RoHS program reduces the use of hazardous substances, including lead, mercury and cadmium
- Two programmable speed dials with emergency and guest service icons on handset
- One non-removable, programmable message retrieval speed dial on handset
- Visual message waiting indicator alerts guests of new messages.Compatible with all major PBXs
- Mute button
- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
- 2-year limited warranty

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.



https://cdn-web.vtp-media.com/products/LS/LS-S3410-USB/LS-S34X0_UG_V6_20210223.pdf

Draw Read aloud 12 of 22

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

The cordless handset and the telephone base cannot be used on the same call. However, calls can be switched between the cordless handset and the telephone base speakerphone.

When the cordless handset or the telephone base is in use, the **TALK** key on the cordless handset and the **LINE** key on the telephone base illuminate.

Receive a call

When there is an incoming call, the telephone rings. The **LINE** key and the **MESSAGE WAITING** LED on the telephone base flash, and the **TALK** key on the cordless handset flashes.

To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or **/SPEAKER.** The **TALK** key illuminates when phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or **/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

96. On information and belief, one or more components of the VTech System comprises one or more processors configured to control playback of the requested media stream without effecting receipt of the requested media stream.

snom

High-resolution color display and self-labeling keys

D717

Key Features

- High-resolution color display
- 3 configurable self-labeling multicolor LED keys
- Wideband audio
- USB port
- Elegant design
- 3-year standard warranty

The D717 SIP Color Deskset features an intuitive design with advanced technology and connectivity. Three self-labeling programmable feature keys sit next to a 2.8-inch, 320 x 240 pixel color LCD display. Opus codec support gives you the freedom to enjoy crystal-clear audio or high-quality narrowband audio, depending on your network conditions. Zero touch provisioning eliminates the need for manual labor typically involved with installation. Power over Ethernet (PoE) support and dual Ethernet ports makes the system cost-effective and easy to use. The USB port allows additional connectivity, including up to three D7 or D7C expansion modules for increased visibility and one-button options. The system features multiple configurations, including three programmable feature keys, four soft keys and 14 dedicated feature keys. You'll feel safe with powerful security features like VPN support.

Product Highlights

- High-resolution 2.8" graphical TFT display
- 3 self-labeling function keys
- Wideband hands-free talking (speakerphone)
- Digital Signal Processor (DSP) enhanced audio quality
- 2-port Gigabit Ethernet switch (RJ45)
- Power over Ethernet IEEE 802.3af, Class 2
- USB headset[1] ready
- D7 or D7C Expansion Module[1] ready
- Support for USB WiFi stick
- Electronic Hook Switch (EHS)[1] support for wireless headsets
- Dual-angle footstand: 46° and 28°

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

Phone Features

- 6 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy / unreachable)[2]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[2]
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[2]
- Call park, call unpark[2]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[2]
- Unified Communications ready

Audio

- Codecs:

Specifications

Protocols

- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface

- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 3 programmable line / function keys with multicolor LEDs
- Dedicated keys for: Message, DND, directory, menu, transfer, hold, page
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 5-way navigation key
- OK and cancel keys
- Sensor-assisted user interface

Security

- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server / client
- Password-protected web interface
- VPN support
- VLAN (IEEE 802.1Q)
- LLDP-MED, RTCP-XR

See e.g., https://www.snomamericas.com/assets/a18da812-db74-426e-ba99-8ebe8ee23621/snom_D717_datasheet_en_20210210.pdf.

PHONE FEATURES

- LS-S342HS requires base phone: LS-S3420-USB
- Environmentally friendly RoHS program reduces the use of hazardous substances, including lead, mercury and cadmium
- Two programmable speed dials with emergency and guest service icons on handset
- One non-removable, programmable message retrieval speed dial on handset
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- 2 lines with line key indication
- Automatic line selection
- Multi-step volume control for ring tones and handset
- Hearing-aid compatible
- 2-year limited warranty

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

Telephone operation

Color SIP Cordless Phone - LS-S3410/LS-S3410-USB

Using the cordless handset and the telephone base

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When the cordless handset or the telephone base is in use, the **TALK** key on the cordless handset and the **LINE** key on the telephone base illuminate.

Receive a call

When there is an incoming call, the telephone rings. The **LINE** key and the **MESSAGE WAITING** LED on the telephone base flash, and the **TALK** key on the cordless handset flashes.

To answer a call using the cordless handset while it is not on the telephone base, charging base or charging stand:

On the cordless handset, press **TALK** or **/SPEAKER.** The **TALK** key illuminates when phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the cordless handset while it is cradled on the telephone base, charging base or charging stand:

Lift the cordless handset from the telephone base, charging base or charging stand. The **TALK** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

To answer a call using the telephone base:

On the telephone base, press **LINE** or **/SPEAKER.** The **LINE** key illuminates when the phone line is in use. The **/SPEAKER** key illuminates when in speakerphone mode.

See e.g., https://www.VTechhotelphones.com/pd/4206/LS-S342HS-2-Line-Color-SIP-Cordless-Accessory-Handset#downloads.

97. ThinkLogix has been damaged by and has suffered irreparable harm as a result of Vtech's infringement.

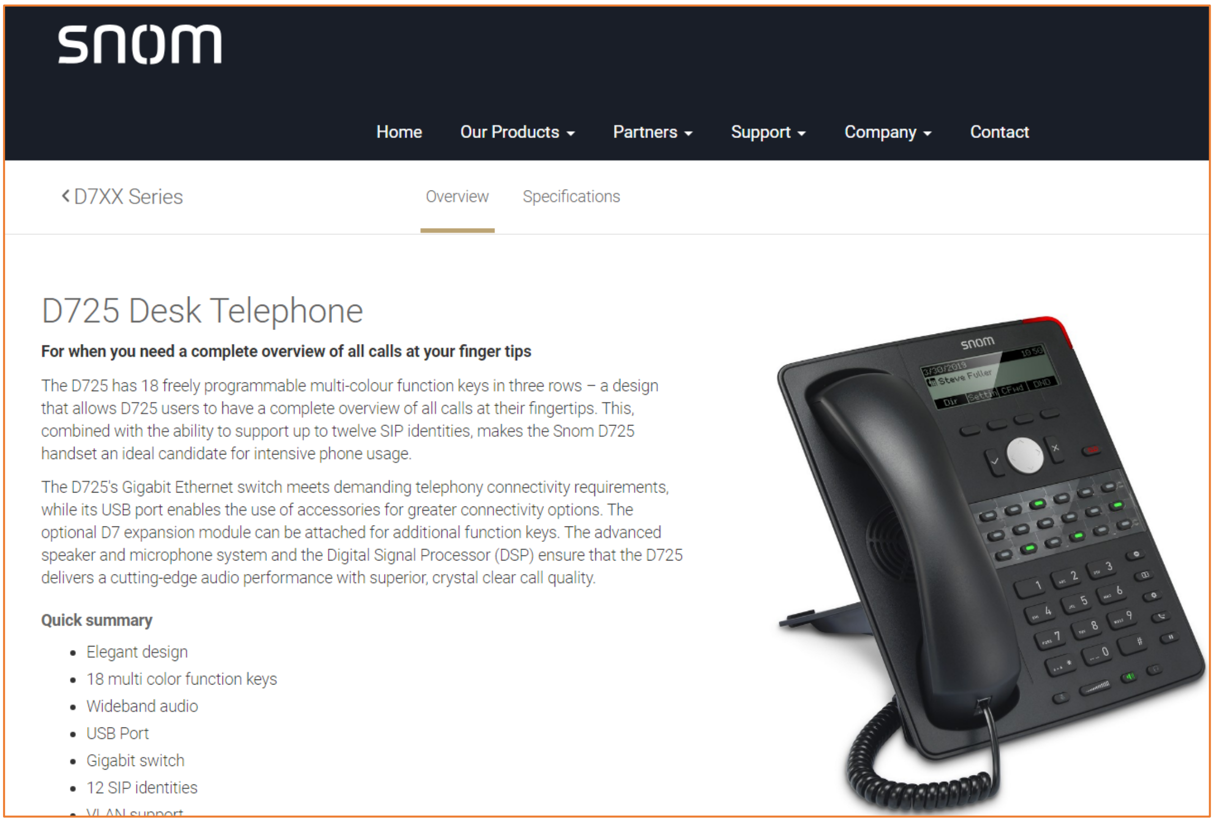
Count V – Infringement of United States Patent No. 7,184,524

98. ThinkLogix repeats, realleges, and incorporates by reference, as if fully set forth here, the allegations of the preceding paragraphs above.

99. On information and belief, VTech is in violation 35 U.S. C. § 271(a) with respect to one or more claims of the ’524 patent.

100. On information and belief, VTech (or those acting on its behalf) has made, used, offered to sell, sold, and/or imported one or more components of the VTech Products and Services that infringes (literally and/or under the doctrine of equivalents) at least claim 1 of the ’524 patent.

101. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for supporting real-time communication (*e.g.*, voice call) between at least two users (*e.g.*, caller and a contact to which the call is requested to be transferred) comprising the step of receiving at least one request (*e.g.*, request to transfer a call) for at least one real-time action (*e.g.*, establishing a voice call), wherein said request (*e.g.*, request to transfer a call) includes a list of participant users (*e.g.*, caller, and a contact, configured with a presence key) to participate in said real-time action (*e.g.*, establishing a voice call).



snom

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‹ D7XX Series Overview Specifications

D725 Desk Telephone

For when you need a complete overview of all calls at your finger tips

The D725 has 18 freely programmable multi-colour function keys in three rows – a design that allows D725 users to have a complete overview of all calls at their fingertips. This, combined with the ability to support up to twelve SIP identities, makes the Snom D725 handset an ideal candidate for intensive phone usage.

The D725's Gigabit Ethernet switch meets demanding telephony connectivity requirements, while its USB port enables the use of accessories for greater connectivity options. The optional D7 expansion module can be attached for additional function keys. The advanced speaker and microphone system and the Digital Signal Processor (DSP) ensure that the D725 delivers a cutting-edge audio performance with superior, crystal clear call quality.

Quick summary

- Elegant design
- 18 multi color function keys
- Wideband audio
- USB Port
- Gigabit switch
- 12 SIP identities
- VLAN support

See e.g., https://www.snomamericas.com/en/pd/d725.

Phone Features

- 12 SIP identities / accounts
- XML browser
- Call lists for dialed, received, missed calls
- Local directory with 1000 entries
- Multiple language support
- DTMF in-band / out-of-band / SIP-INFO
- Interoperable with all major IP-PBX platforms
- Speed dialing
- URL dialing
- Local dial plan
- Automatic redial on busy
- Call completion (busy/unreachable)[3]
- Caller identification
- Call waiting
- Call blocking (deny list)
- Auto answer
- Hold
- Music on hold[3]
- Handling of up to 12 simultaneous calls
- Blind and attended transfer
- Call forwarding
- 3-way conferencing on the phone
- Extension monitoring, call pickup[3]
- Call park, call unpark[3]
- Multicast paging
- DND mode (do not disturb)
- Keyboard lock
- Client matter code (CMC)[3]
- Unified communications ready

Audio
- Codecs:
  - G.711 A-law, μ-law
  - G.722 (wideband)
  - G.726, G.729AB, GSM 6.10 (FR)
- Built-in assignable ringtones
- Comfort noise generator (CNG)
- Voice activity detection (VAD)

Specifications

Protocols
- SIP (RFC3261)
- DHCP, NTP
- HTTP / HTTPS / TFTP
- LDAP (Directory)
- Dual Stack IPv4 / IPv6

User Interface
- Localization (language, time, dial tone)
- Red LED for call indication / message waiting
- 4 context sensitive keys
- 18 programmable line / function keys with multicolor LEDs
- Paper inlay for labeling of function keys
- Dedicated keys for: Message, DND, directory, menu, transfer, hold
- Audio keys with LED indication: Mute, speakerphone, headset
- Volume key
- 4-way navigation key, OK and cancel keys
- Menu-driven user interface

Security
- 802.1X Authentication and EAPOL
- Transport layer security (TLS)
- SRTP (RFC3711), SIPS, RTCP
- HTTPS server/client
- Password-protected web interface
- VPN support
- VLAN (IEEE 802.1Q)
- LLDP-MED, RTCP-XR

Specifications, Weights and Dimensions
- Dimensions (approx.):
  - 7.48" x 3.54" x 9.37"
  - With desk stand: 7.48 x 8.6" x 7.25"
- Weight (approx.): 1.8 lbs
- Input voltage: 5V DC (SELV)
- Power supply: PoE or power adapter
- Environmental conditions:
  - Ambient temperature: 0°C to 35°C (32°F to 95°F)
  - Storage temperature: -10°C to 45°C (14°F to 113°F)
  - Humidity: 5% to 95% (non-condensing)
- Included in delivery:
  - Base phone unit
  - Foot stand
  - Handset with cord
  - Ethernet cable
  - Quick Start Guide
- Color: Black
- Warranty: 3 years
- Part Number: 00003916 (all markets)
- APN: 80-S003-00

[1] Optional accessory
[2] Available separately
[3] If supported by PBX

APN numbers for optional accessories

80-S007-00	D7 Expansion Module (black)
26-350200-3UL	Power Adapter Snom A6 / US clip
89-S015-00	Snom EHS Advanced
00-S013-00	Wallmount plate for 7-series
89-4083-00	A100D Binaural headset
89-4082-00	A100M Mono headset
80-S058-00	A210 Wi-Fi USB Dongle
80-S050-00	A230 DECT USB Dongle
80-S094-00	A171 DECT Headset

See e.g., https://www.snomamericas.com/en/pd/d725.

Integration with Snom

Since its launch in 2010, Snom has been member of the "Compatible with Microsoft Lync (3PIP)" program by Microsoft. The *Snom UC edition* features a complete portfolio of IP phones 100% qualified for and compatible with Microsoft Lync. Snom has an easily deployable phone to meet any business requirement.

Phone Model	D765	D725	D715	760/720/710	300/821
Qualified for Lync 2013	●	●	●	●	
Qualified for Lync 2010	●	●	●	●	●

See e.g., https://www.snomamericas.com/eco-system/pbx-interoperability-partner/microsoft.

Presence	If supported by your PBX, the LED of this function key will reflect the presence status (ringing, busy, available, etc.) of the extension specified in the "Number" text field. The function key can also be used to dial the extension, usually when the destination signals availability.

Presence keys

You can put the Presence function onto one of the function keys (see diagram on page 10) and associate that key to one of your contacts. When your contact is available, the LED of the key will signal the availability with a green light; when your contact is in a call or otherwise unavailable, the LED signals the unavailability with a red light.

- Presence status: Available

- Presence status: Unavailable (inactive, away, etc.)

- Presence status: Busy (in call, in a meeting, etc.)

Using Presence function keys

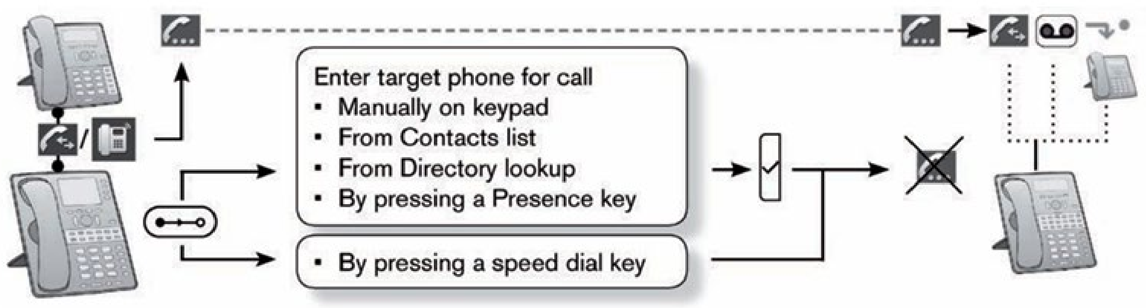
You can call your contact by simply pressing the key twice, and you can use the key to transfer calls to your contact (see "Transferring calls to Presence keys - attended" on page 31, "Transferring calls to Presence keys - unattended" on page 36, and "Transferring calls to Presence keys safely" on page 41).

Transferring calls

You can transfer connected calls as well as calls ringing on your phone.

- When you have a call on the line, there are two ways to transfer it to a third party:
  - Announcing the call to the third party first, to make sure the call is welcome and will be accepted: Attended transfer;
  - Transferring the call unannounced: Blind transfer. There will be no feedback on whether the third party is available and/or picking up the call.
- When a call is ringing on your phone, you can transfer it to a third party without answering it first (blind transfer only).

Unattended (blind) transfers



1. With a call ringing or an active call on the line, press . The active call is put on hold, and you will see the dial screen.

2. Press the Presence key . The Presence screen appears. "Transfer to contact" is the default selection when the screen opens. If you want to call one of the other available numbers, use ▲ / ▼ to scroll to it.

3. Press . While the phone is calling the number, you will either see the the hold screen (if you are transferring an active call) or a new missed call (if you are transferring a ringing call).

See e.g., from Datasheet of accused product.

102. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for supporting real-time communication (*e.g.*, voice call) between at least two users (*e.g.*, caller and a contact to which the call is requested to be transferred) comprising in a first determining step, determining, responsive to said at least one request (*e.g.*, request to transfer a call), a status (*e.g.*, available, busy, away etc.) of at least one condition (*e.g.*, presence of a contact, configured with a presence key), wherein said status (*e.g.*, available, busy, away etc.) of said at least one condition (*e.g.*, presence of a contact, configured with a presence key) indicates whether at least one of said participant users (*e.g.*, caller and a contact configured with a presence key) is available to participate in said real-time action (*e.g.*, establishing a voice call), and wherein said status (*e.g.*, available, busy, away etc.) of said at least one condition

(*e.g.*, presence of a contact configured with a presence key) is responsive to an online presence (*e.g.*, led indicating status of the contact configured with a presence key) of said at least one of said participant users (*e.g.*, caller and a contact configured with a presence key).

Presence	If supported by your PBX, the LED of this function key will reflect the presence status (ringing, busy, available, etc.) of the extension specified in the "Number" text field. The function key can also be used to dial the extension, usually when the destination signals availability.

Presence keys

You can put the Presence function onto one of the function keys (see diagram on page 10) and associate that key to one of your contacts. When your contact is available, the LED of the key will signal the availability with a green light; when your contact is in a call or otherwise unavailable, the LED signals the unavailability with a red light.

- Presence status: Available

- Presence status: Unavailable (inactive, away, etc.)

- Presence status: Busy (in call, in a meeting, etc.)

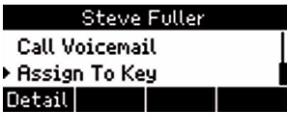
Using Presence function keys

You can call your contact by simply pressing the key twice, and you can use the key to transfer calls to your contact (see "Transferring calls to Presence keys - attended" on page 31, "Transferring calls to Presence keys - unattended" on page 36, and "Transferring calls to Presence keys safely" on page 41).

Configuring Presence function keys for Contacts

1. Press Contac and use ▲ / ▼ to select the Contact.

2. Press Detail.



3. Use ▲ / ▼ to scroll to **Assign to Key** and press .

4. To select a key, just enter the respective number on the keypad. Alternatively, you can also use ▲ / ▼ to scroll to the key , then press .

Transferring calls to Presence keys - unattended

1. With a call ringing or an active call on the line, press . The active call is put on hold, and you will see the dial screen.

2. Press the Presence key . The Presence screen appears. "Transfer to contact" is the default selection when the screen opens. If you want to call one of the other available numbers, use ▲ / ▼ to scroll to it.

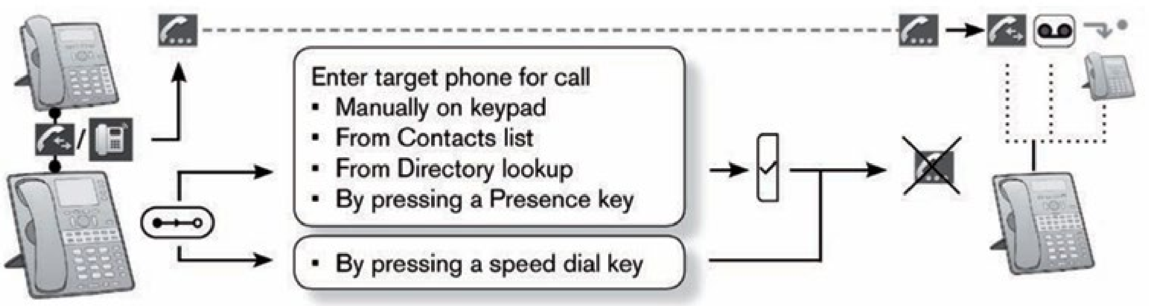
3. Press . While the phone is calling the number, you will either see the the hold screen (if you are transferring an active call) or a new missed call (if you are transferring a ringing call).

Transferring calls

You can transfer connected calls as well as calls ringing on your phone.

- When you have a call on the line, there are two ways to transfer it to a third party:
  - Announcing the call to the third party first, to make sure the call is welcome and will be accepted: Attended transfer;
  - Transferring the call unannounced: Blind transfer. There will be no feedback on whether the third party is available and/or picking up the call.
- When a call is ringing on your phone, you can transfer it to a third party without answering it first (blind transfer only).

Unattended (blind) transfers



See e.g., from Datasheet of accused product.

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Presence	If supported by your PBX, the LED of this function key will reflect the presence status (ringing, busy, available, etc.) of the extension specified in the "Number" text field. The function key can also be used to dial the extension, usually when the destination signals availability.

Presence keys

You can put the Presence function onto one of the function keys (see diagram on page 10) and associate that key to one of your contacts. When your contact is available, the LED of the key will signal the availability with a green light; when your contact is in a call or otherwise unavailable, the LED signals the unavailability with a red light.

- Presence status: Available

- Presence status: Unavailable (inactive, away, etc.)

- Presence status: Busy (in call, in a meeting, etc.)

Using Presence function keys

You can call your contact by simply pressing the key twice, and you can use the key to transfer calls to your contact (see "Transferring calls to Presence keys - attended" on page 31, "Transferring calls to Presence keys - unattended" on page 36, and "Transferring calls to Presence keys safely" on page 41).

Configuring Presence function keys for Contacts

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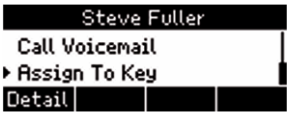


3. Use ▲ / ▼ to scroll to **Assign to Key** and press .

4. To select a key, just enter the respective number on the keypad. Alternatively, you can also use ▲ / ▼ to scroll to the key , then press .

Transferring calls to Presence keys - unattended

1. With a call ringing or an active call on the line, press . The active call is put on hold, and you will see the dial screen.

2. Press the Presence key . The Presence screen appears. "Transfer to contact" is the default selection when the screen opens. If you want to call one of the other available numbers, use ▲ / ▼ to scroll to it.

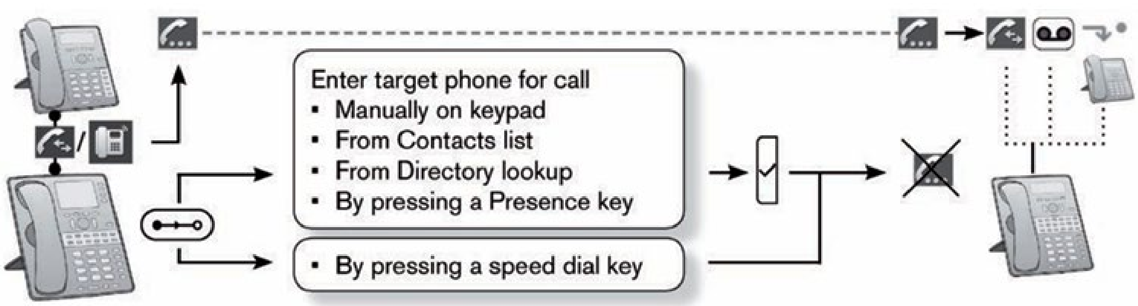
3. Press . While the phone is calling the number, you will either see the the hold screen (if you are transferring an active call) or a new missed call (if you are transferring a ringing call).

Transferring calls

You can transfer connected calls as well as calls ringing on your phone.

- When you have a call on the line, there are two ways to transfer it to a third party:
  - Announcing the call to the third party first, to make sure the call is welcome and will be accepted: Attended transfer;
  - Transferring the call unannounced: Blind transfer. There will be no feedback on whether the third party is available and/or picking up the call.
- When a call is ringing on your phone, you can transfer it to a third party without answering it first (blind transfer only).

Unattended (blind) transfers



See e.g., from Datasheet of accused product.

104. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for supporting real-time communication (*e.g.*, voice call) between at least two users (*e.g.*, caller and a contact to which the call is requested to be transferred) comprising the step of performing said real-time action (*e.g.*, establishing a voice call) in the event that a determination is made that said real-time action (*e.g.*, establishing a voice call) can be performed.

Transferring calls to Presence keys - unattended

1. With a call ringing or an active call on the line, press . The active call is put on hold, and you will see the dial screen.

2. Press the Presence key . The Presence screen appears. "Transfer to contact" is the default selection when the screen opens. If you want to call one of the other available numbers, use ▲ / ▼ to scroll to it.

3. Press . While the phone is calling the number, you will either see the the hold screen (if you are transferring an active call) or a new missed call (if you are transferring a ringing call).

Transferring calls

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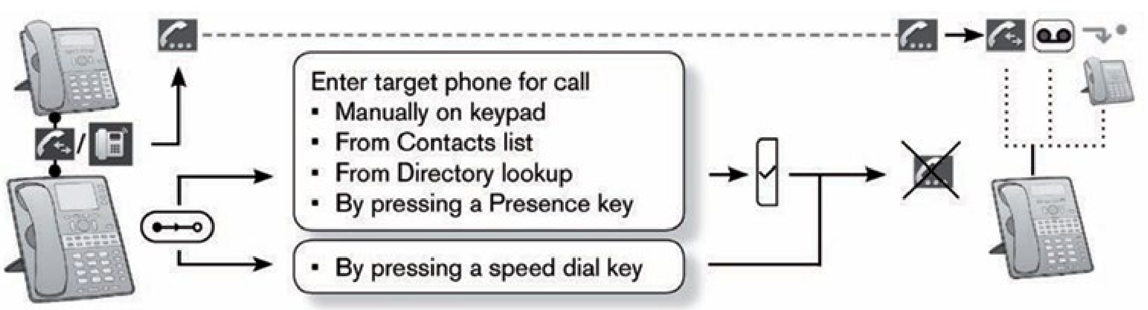
Unattended (blind) transfers



See e.g., from Datasheet of accused product.

105. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for supporting real-time communication (*e.g.*, voice call) between at least two users (*e.g.*, caller and a contact to which the call is requested to be transferred) comprising the step of wherein said second determining step and said performing step are responsive to a processing of at least one rule (*e.g.*, mandating presence of contact indicated by green led and the contact picking up a call), said at least one rule (*e.g.*, mandating presence of contact indicated by green led and the contact picking up a call), having a logical structure defined by at least one of at least one condition (*e.g.*, presence of a contact configured with a presence key) and at least one event

notification (*e.g.*, if the contact, configured with a presence key, do not pick up a call after a time), said at least one rule (*e.g.*, mandating presence of contact indicated by green led and the contact picking up a call), defining a specified relationship between said at least one condition (*e.g.*, presence of a contact configured with a presence key), said at least one event notification (*e.g.*, if the contact, configured with a presence key, do not pick up a call after a time), and said real-time action (*e.g.*, establishing a voice call).

Call forwarding

The phone can be set to forward incoming calls, either always or under certain conditions. The settings can be done on the phone and on the web interface. For further information on using the web interface for these settings, see "Call forwarding" on page 71.

Forward All: Forwarding all incoming calls to the number of the phone, extension, or mailbox specified as this function's target.

As the default setting, the function key CFwd in the function key line turns forwarding of all calls on and off, but you can also map the function onto another function key or use the settings menu as shown in the table, below.

Forward when Busy: Forwarding calls ringing while phone is busy to the number of the phone, extension, or mailbox specified as this function's target.

Forward after Timeout: When a call starts ringing, the phone will wait for the number of seconds specified in the setting "Call forwarding time". If the call is not accepted by the end of this time period, it is forwarded to the number of the phone, extension, or mailbox specified as this function's target.

Presence	If supported by your PBX, the LED of this function key will reflect the presence status (ringing, busy, available, etc.) of the extension specified in the "Number" text field. The function key can also be used to dial the extension, usually when the destination signals availability.

Presence keys

You can put the Presence function onto one of the function keys (see diagram on page 10) and associate that key to one of your contacts. When your contact is available, the LED of the key will signal the availability with a green light; when your contact is in a call or otherwise unavailable, the LED signals the unavailability with a red light.

- Presence status: Available

- Presence status: Unavailable (inactive, away, etc.)

- Presence status: Busy (in call, in a meeting, etc.)

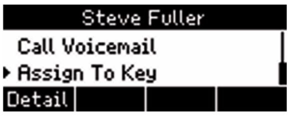
Using Presence function keys

You can call your contact by simply pressing the key twice, and you can use the key to transfer calls to your contact (see "Transferring calls to Presence keys - attended" on page 31, "Transferring calls to Presence keys - unattended" on page 36, and "Transferring calls to Presence keys safely" on page 41).

Configuring Presence function keys for Contacts

1. Press Contac and use ▲ / ▼ to select the Contact.

2. Press Detail.



3. Use ▲ / ▼ to scroll to **Assign to Key** and press .

4. To select a key, just enter the respective number on the keypad. Alternatively, you can also use ▲ / ▼ to scroll to the key , then press .

Transferring calls to Presence keys - unattended

1. With a call ringing or an active call on the line, press . The active call is put on hold, and you will see the dial screen.

2. Press the Presence key . The Presence screen appears. "Transfer to contact" is the default selection when the screen opens. If you want to call one of the other available numbers, use ▲ / ▼ to scroll to it.

3. Press . While the phone is calling the number, you will either see the the hold screen (if you are transferring an active call) or a new missed call (if you are transferring a ringing call).

4. When your contact picks up the ringing call (or if the call is forwarded to the third party's VoiceMail or another phone number), you will see the message at Fig. 1:

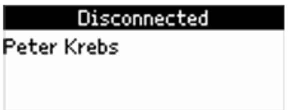


Fig. 1

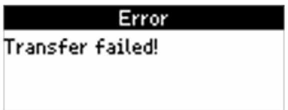


Fig. 2

Fig. 3

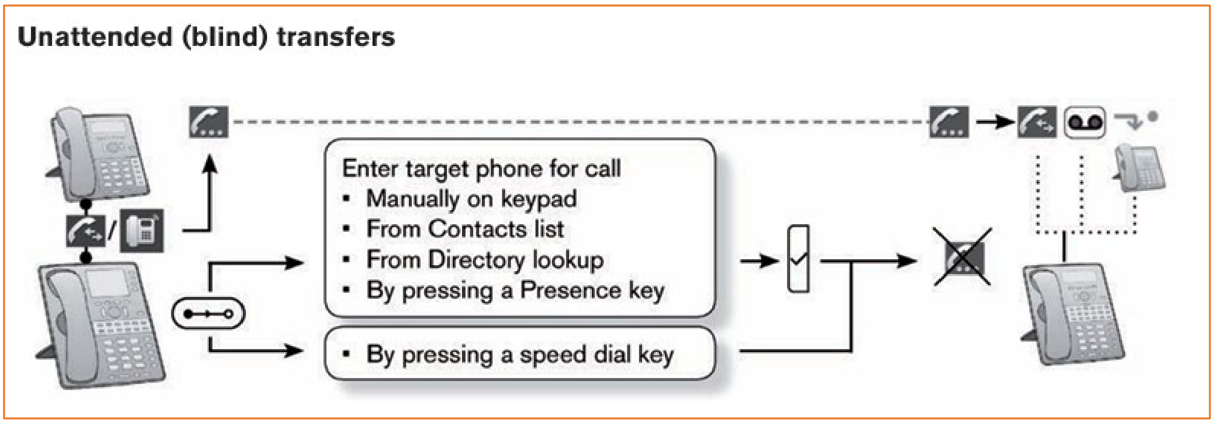
If the call is not picked up or forwarded, you will hear a double beep and see the error screen (Fig. 2). Then the caller will be reconnected to your phone (Fig. 3).

Transferring calls

You can transfer connected calls as well as calls ringing on your phone.

- When you have a call on the line, there are two ways to transfer it to a third party:
  - Announcing the call to the third party first, to make sure the call is welcome and will be accepted: Attended transfer;
  - Transferring the call unannounced: Blind transfer. There will be no feedback on whether the third party is available and/or picking up the call.

- When a call is ringing on your phone, you can transfer it to a third party without answering it first (blind transfer only).



See e.g., from Datasheet of accused product

106. ThinkLogix has been damaged by and has suffered irreparable harm as a result of Vtech's infringement.

<u>Count VI – Infringement of United States Patent No. 7,136,392</u>

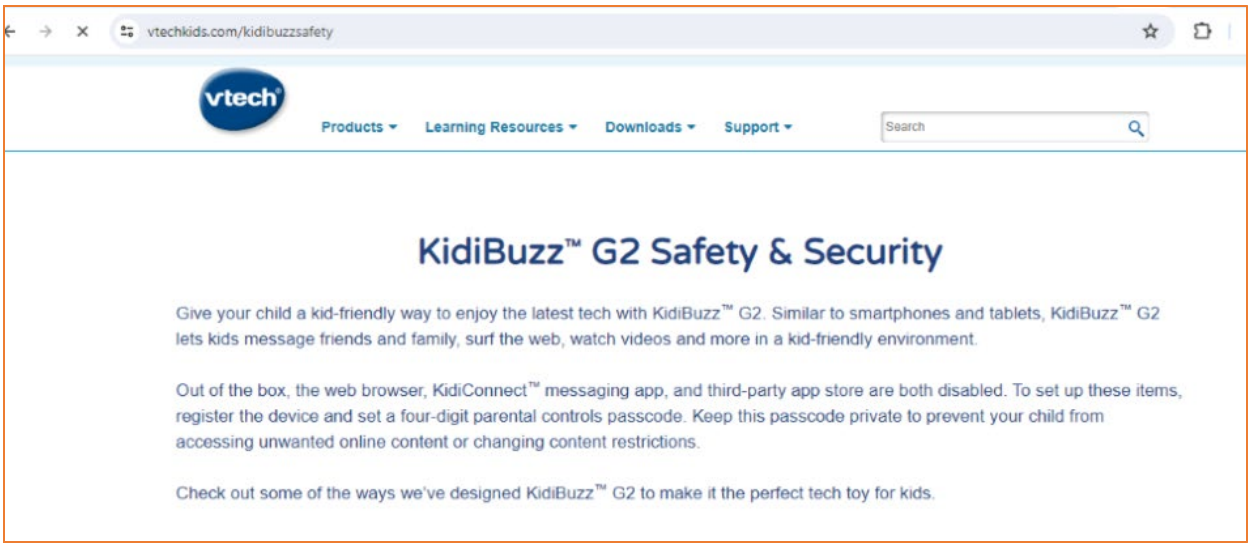
107. ThinkLogix repeats, realleges, and incorporates by reference, as if fully set forth here, the allegations of the preceding paragraphs above.

108. On information and belief, VTech is in violation 35 U.S. C. § 271(a) with respect to one or more claims of the '392 patent.

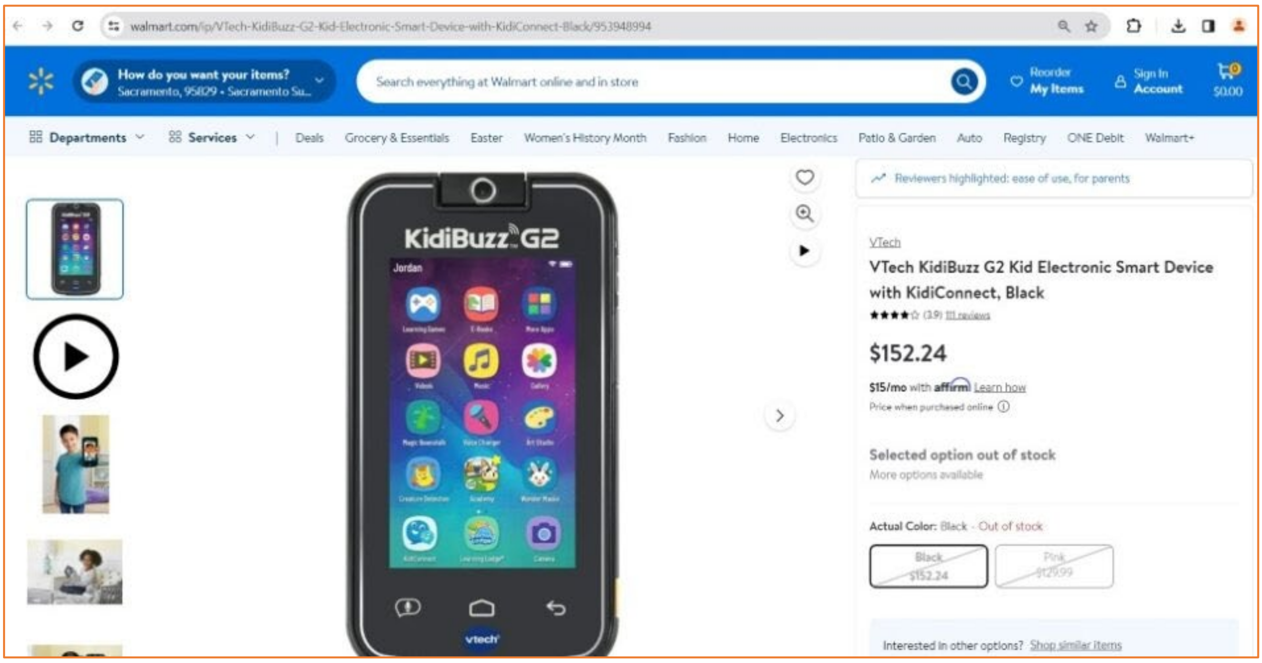
109. On information and belief, VTech (or those acting on its behalf) has made, used, offered to sell, sold, and/or imported one or more components of the VTech Products and Services that infringes (literally and/or under the doctrine of equivalents) at least claim 4 of the '392 patent.

110. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for comprising the step of directing to a first output queue (e.g., an output queue of an access category, i.e., background category queue, of an WMM STA or AP such as the accused product) at a first station (e.g., transmitting STA or AP such as the accused product) of a communication network (e.g., WLAN network), message data

units (e.g., MPDU, MSDU, MMPDU, etc.) to be transmitted over a communication medium (e.g., wireless medium) and having a first traffic classification (e.g., access categories such as background traffic, etc.).



See e.g., https://www.vtechkids.com/kidibuzzsafety.



See e.g., https://www.walmart.com/ip/VTech-KidiBuzz-G2-Kid-Electronic-Smart-Device-with-KidiConnect-Black/953948994.

Certification ID: WFA83894

Date of Last Certification: Jun 20, 2019
Brand: VTech Electronics Ltd.
Category: Gaming Device - Portable
Product Name: KidiBuzz G2
Model Number: 80-1866
Total Variants: 1

Variant #1 of 1 matches

Date of Certification: Jun 20, 2019
Product Model Variant: 2019-06-20 (WFA83894 - 8358009)
Operating System: Android, version:6.0
Frequency Band(s): 2.4 GHz; 5 GHz

Summary of Certifications for Variant #1

CLASSIFICATION	PROGRAM
Spectrum & Regulatory Features	Spectrum & Regulatory
Connectivity	Wi-Fi CERTIFIED™ n
	Wi-Fi CERTIFIED™ a
	Wi-Fi CERTIFIED™ b
	Wi-Fi CERTIFIED™ g
	2.4 GHz Spectrum Capabilities
	5 GHz Spectrum Capabilities
Optimization	WMM®

See e.g., https://www.wi-fi.org/product-finder-results?keywords=WFA83894.

1 Overview

1.1 Purpose of This Document

This document defines the specification for WMM, an 802.11 quality of service (QoS) implementation based on a subset of the draft 802.11e standard supplement [2]. It is motivated by the need to prevent market fragmentation caused by multiple, non-interoperable pre-standard subsets of the draft 802.11e standard that would otherwise occur. It is intended that WMM can be implemented, subjected to interoperability testing and deployed in the market before the availability of 802.11e. This is facilitated by selecting a subset of the features of 802.11e. In no way should WMM be taken to detract from 802.11e itself, which is viewed as the long term endpoint of WMM. Deployment of WMM will deliver useful QoS functionality for voice over 802.11, streaming media and also provide key lessons which will benefit eventual deployment of 802.11e.

1.3 WMM Features

The features supported by WMM in this phase are as follows:

1. Capability negotiation independent of 802.11e. That is, WMM devices will not advertise 802.11e capability unless they also support those features independently. This is part of a forward compatibility strategy which is described in detail in a subsequent paragraph.
2. Frame formats and over the air protocols will be based on those currently proposed for 802.11e. However, no attempt will be made to track future changes in 802.11e and reflect them back into the WMM specification. Divergence between the two specifications is a necessary side effect of the need to freeze the WMM specification as soon as possible.
3. WMM will use an EDCF mechanism only, and except where explicitly indicated otherwise in this specification other 802.11e features, including HCF polling and associated signaling, Block Acknowledgement, and side traffic, are not part of WMM.
4. Interfaces to the MAC which signal per-packet priority will be consistent with those used for Ethernet, both in terms of the driver API and bridging to other 802 link layers via an 802.1D bridge.
5. The number of exposed queues will be fixed at four. A fixed mapping of priority information carried in the 802.1D Priority field to those four queues will be defined, together with suggested uses for each priority consistent with the suggested uses in 802.1D.

See e.g., screen captures from WMM Specification Version 1.1.

This amendment defines the medium access control (MAC) procedures to support local area network (LAN) applications with quality of service (QoS) requirements, including the transport of voice, audio, and video over IEEE 802.11 wireless LANs (WLANs).

See e.g., screen captures from the IEEE 802-11e Standard.

3.82 medium access control (MAC) protocol data unit (MPDU): The unit of data exchanged between two peer MAC entities using the services of the physical layer (PHY).

3.83 medium access control (MAC) service data unit (MSDU): Information that is delivered as a unit between MAC service access points (SAPs).

3.57 fragmentation: The process of segmenting a medium access control (MAC) service data unit (MSDU) or MAC management protocol data unit (MMPDU) into a sequence of smaller MAC protocol data units (MPDUs) prior to transmission. The process of recombining a set of fragment MPDUs into an MSDU or MMPDU is known as defragmentation. These processes are described in 5.8.1.9 of ISO/IEC 7498-1:1994.

See e.g., screen captures from the IEEE 802-11 Standard.

7.1 MAC frame formats

Change the text of 7.1 as follows:

Each frame consists of the following basic components:

a) A *MAC header*, which comprises frame control, duration, address, ~~and~~ sequence control information, and, for QoS data frames, QoS control information;

b) A variable length *frame body*, which contains information specific to the frame *type* and *subtype*;

c) A *frame check sequence* (FCS), which contains an IEEE 32-bit CRC.

Octets: 2	2	6	6	6	2	6	2	0-23~~12~~24	4
Frame Control	Duration/ ID	Address 1	Address 2	Address 3	Sequence Control	Address 4	QoS Control	Frame Body	FCS

MAC Header

Figure 12—MAC frame format

See e.g., screen captures from the IEEE 802-11e Standard.

2.1.6 QoS Control Field

The QoS control field consists of two octets and is shown in Figure 4.

15 7	6 5	4	3	2 0
0	ack policy	EOSP	0	UP

Figure 4 QoS Control Field

The three bit UP field carries the priority bits of the 802.1D Priority and is used to signal the priority for this frame. It also implies the sending AC according to the mappings in Table 14. The UP for each MPDU of a MSDU shall be the same value.

3.3 Assignment of Frames to Queues

3.3.1 Mappings for Unicast Frames

The MAC data service at a STA or AP provides for connectionless, asynchronous transport of MSDUs. Each MSDU transfer request includes an 802.1D Priority field equal to that value. The priority bits of the 802.1D field are mapped to Access Category (AC) according to Table 14 and are listed in increasing priority order. The UP field is carried in the QoS control field of an MPDU. The UP field references the AC the MPDU is transmitted at using the mapping defined in Table 14. At the receiver, the UP field carried in the MPDU shall be used to re-create the 802.1D priority information of the MSDU.

Table 14 802.1D Priority to AC mappings

<table>
<tr><th>Priority</th><th>802.1D Priority
(= UP)</th><th>802.1D Designation</th><th>Access Category</th><th>WMM Designation</th></tr>
<tr><td rowspan="8">lowest
↓
highest</td><td>1</td><td>BK</td><td rowspan="2">AC_BK</td><td rowspan="2">Background</td></tr>
<tr><td>2</td><td>-</td></tr>
<tr><td>0</td><td>BE</td><td rowspan="2">AC_BE</td><td rowspan="2">Best Effort</td></tr>
<tr><td>3</td><td>EE</td></tr>
<tr><td>4</td><td>CL</td><td rowspan="2">AC_VI</td><td rowspan="2">Video</td></tr>
<tr><td>5</td><td>VI</td></tr>
<tr><td>6</td><td>VO</td><td rowspan="2">AC_VO</td><td rowspan="2">Voice</td></tr>
<tr><td>7</td><td>NC</td></tr>
</table>

Transmit frames are then placed in queues according to AC. The AP and STA may implement more queues for internal prioritization. Data frames with no priority information are treated as best effort.

Management frames have no QoS Control field, but should be transmitted with parameters of AC_VO

PS-Poll frames should be mapped to AC_BE.

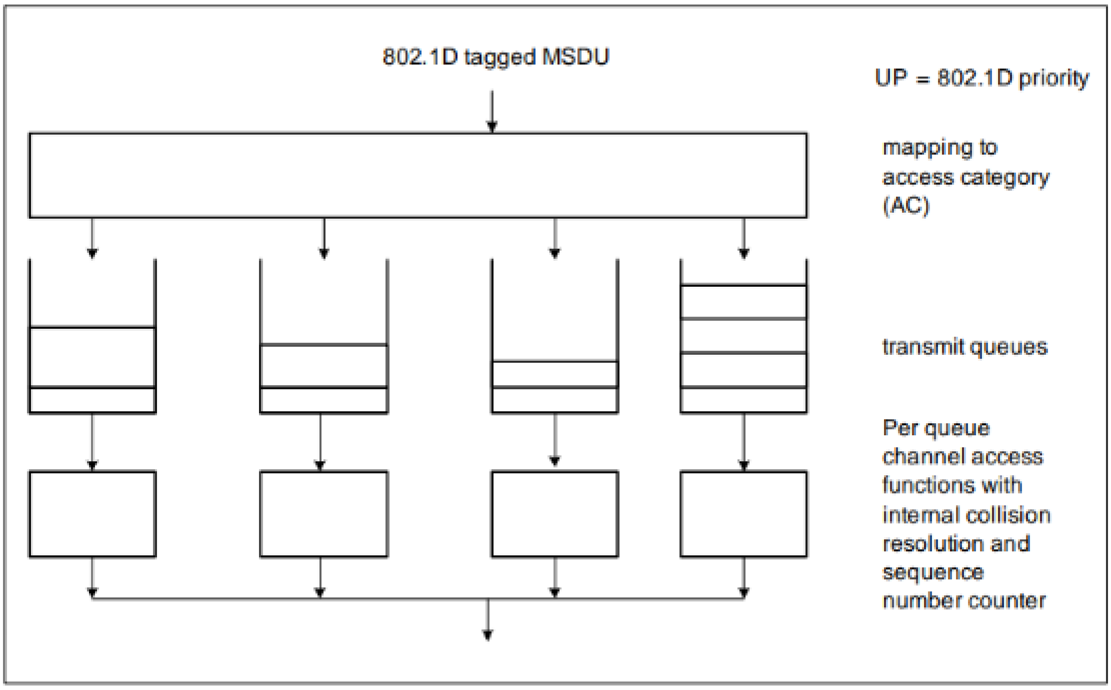
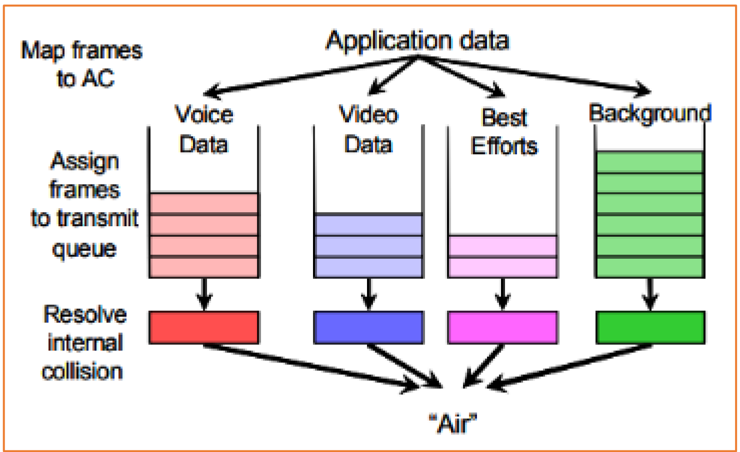


Figure 17 Reference Implementation Model

See e.g., screen captures from the WMM Specification Version 1.1.



See e.g., screen captures from the WiFi Certified for WMM – Support for Multimedia Applications with Quality of Service in Wi-Fi Networks.

111. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for comprising the step of directing to a second output queue (e.g., an output queue of an access category, i.e., voice category queue, of an WMM STA or AP such as the

accused product) at the first station (e.g., transmitting STA or AP such as the accused product), message data units (e.g., MPDU, MMPDU or MSDU, etc.) to be transmitted over the communication medium (e.g., wireless medium) and having a second traffic classification (e.g., voice traffic, etc.).

Certification ID: WFA83894

vtech

Date of Last Certification: Jun 20, 2019
Brand: VTech Electronics Ltd.
Category: Gaming Device - Portable
Product Name: KidiBuzz G2
Model Number: 80-1866
Total Variants: 1

Variant #1 of 1 matches

Date of Certification: Jun 20, 2019
Product Model Variant: 2019-06-20 (WFA83894 - 8358009)
Operating System: Android, version:6.0
Frequency Band(s): 2.4 GHz; 5 GHz

Summary of Certifications for Variant #1

CLASSIFICATION	PROGRAM
Spectrum & Regulatory Features	Spectrum & Regulatory
Connectivity	Wi-Fi CERTIFIED™ n
	Wi-Fi CERTIFIED™ a
	Wi-Fi CERTIFIED™ b
	Wi-Fi CERTIFIED™ g
	2.4 GHz Spectrum Capabilities
	5 GHz Spectrum Capabilities
Optimization	WMM®

See e.g., https://www.wi-fi.org/product-finder-results?keywords=WFA83894.

1 Overview

1.1 Purpose of This Document

This document defines the specification for WMM, an 802.11 quality of service (QoS) implementation based on a subset of the draft 802.11e standard supplement [2]. It is motivated by the need to prevent market fragmentation caused by multiple, non-interoperable pre-standard subsets of the draft 802.11e standard that would otherwise occur. It is intended that WMM can be implemented, subjected to interoperability testing and deployed in the market before the availability of 802.11e. This is facilitated by selecting a subset of the features of 802.11e. In no way should WMM be taken to detract from 802.11e itself, which is viewed as the long term endpoint of WMM. Deployment of WMM will deliver useful QoS functionality for voice over 802.11, streaming media and also provide key lessons which will benefit eventual deployment of 802.11e.

1.3 WMM Features

The features supported by WMM in this phase are as follows:

1. Capability negotiation independent of 802.11e. That is, WMM devices will not advertise 802.11e capability unless they also support those features independently. This is part of a forward compatibility strategy which is described in detail in a subsequent paragraph.
2. Frame formats and over the air protocols will be based on those currently proposed for 802.11e. However, no attempt will be made to track future changes in 802.11e and reflect them back into the WMM specification. Divergence between the two specifications is a necessary side effect of the need to freeze the WMM specification as soon as possible.
3. WMM will use an EDCF mechanism only, and except where explicitly indicated otherwise in this specification other 802.11e features, including HCF polling and associated signaling, Block Acknowledgement, and side traffic, are not part of WMM.
4. Interfaces to the MAC which signal per-packet priority will be consistent with those used for Ethernet, both in terms of the driver API and bridging to other 802 link layers via an 802.1D bridge.
5. The number of exposed queues will be fixed at four. A fixed mapping of priority information carried in the 802.1D Priority field to those four queues will be defined, together with suggested uses for each priority consistent with the suggested uses in 802.1D.

See e.g., screen captures from WMM Specification Version 1.1.

This amendment defines the medium access control (MAC) procedures to support local area network (LAN) applications with quality of service (QoS) requirements, including the transport of voice, audio, and video over IEEE 802.11 wireless LANs (WLANs).

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See e.g., screen captures from the IEEE 802-11 Standard.

7.1 MAC frame formats

Change the text of 7.1 as follows:

Each frame consists of the following basic components:

a) A *MAC header,* which comprises frame control, duration, address, ~~and~~ sequence control information, and, for QoS data frames, QoS control information;

b) A variable length *frame body*, which contains information specific to the frame *type* and *subtype*;

c) A *frame check sequence* (FCS), which contains an IEEE 32-bit CRC.

Octets: 2	2	6	6	6	2	6	2	0-23124	4
Frame Control	Duration/ ID	Address 1	Address 2	Address 3	Sequence Control	Address 4	QoS Control	Frame Body	FCS

MAC Header

Figure 12—MAC frame format

See e.g., screen captures from the IEEE 802-11e Standard.

2.1.6 QoS Control Field

The QoS control field consists of two octets and is shown in Figure 4.

15 7	6 5	4	3	2 0
0	ack policy	EOSP	0	UP

Figure 4 QoS Control Field

The three bit UP field carries the priority bits of the 802.1D Priority and is used to signal the priority for this frame. It also implies the sending AC according to the mappings in Table 14. The UP for each MPDU of a MSDU shall be the same value.

3.3 Assignment of Frames to Queues

3.3.1 Mappings for Unicast Frames

The MAC data service at a STA or AP provides for connectionless, asynchronous transport of MSDUs. Each MSDU transfer request includes an 802.1D Priority field equal to that value. The priority bits of the 802.1D field are mapped to Access Category (AC) according to Table 14 and are listed in increasing priority order. The UP field is carried in the QoS control field of an MPDU. The UP field references the AC the MPDU is transmitted at using the mapping defined in Table 14. At the receiver, the UP field carried in the MPDU shall be used to re-create the 802.1D priority information of the MSDU.

Table 14 802.1D Priority to AC mappings

Priority	802.1D Priority (= UP)	802.1D Designation	Access Category	WMM Designation
lowest	1	BK		
	2	-	AC_BK	Background
	0	BE		
	3	EE	AC_BE	Best Effort
	4	CL		
	5	VI	AC_VI	Video
	6	VO		
highest	7	NC	AC_VO	Voice

Transmit frames are then placed in queues according to AC. The AP and STA may implement more queues for internal prioritization. Data frames with no priority information are treated as best effort.

Management frames have no QoS Control field, but should be transmitted with parameters of AC_VO

PS-Poll frames should be mapped to AC_BE.

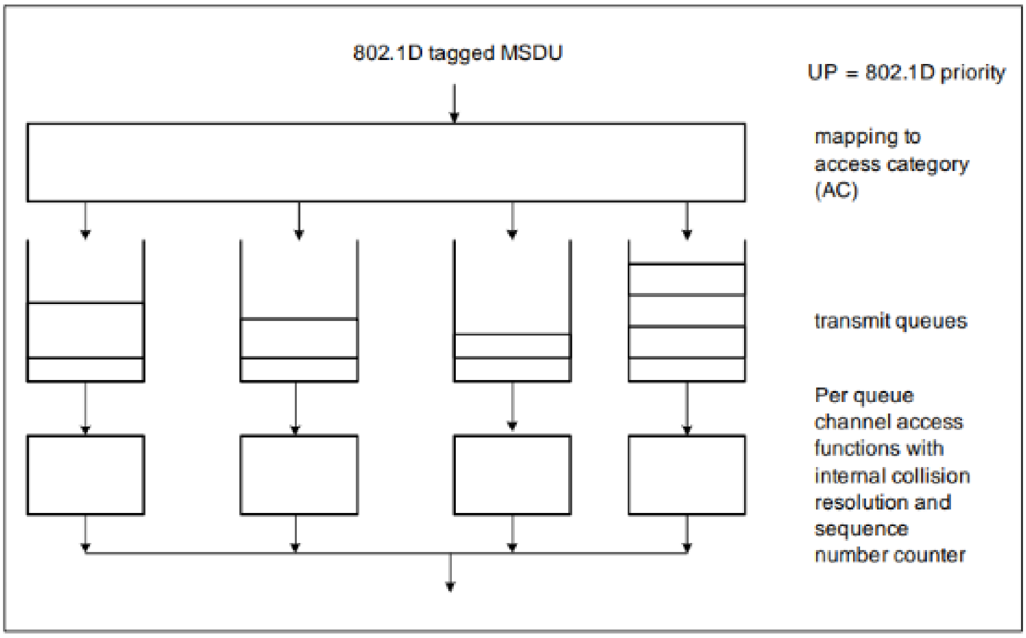
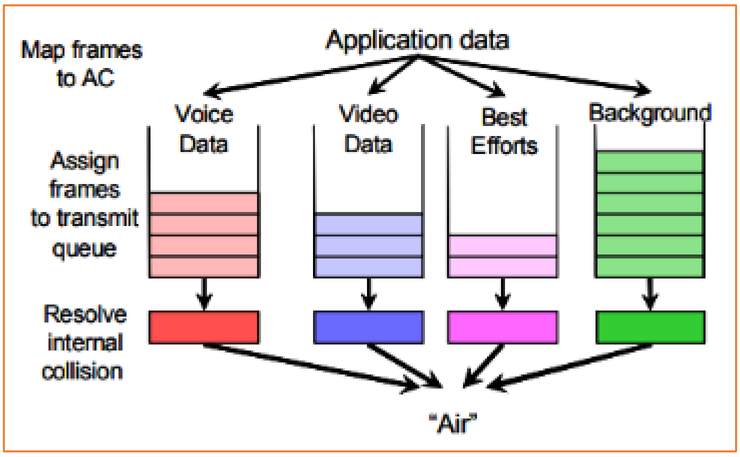


Figure 17 Reference Implementation Model

See e.g., screen captures from the WMM Specification Version 1.1.



See e.g., screen capture from the WiFi Certified for WMM – Support for Multimedia Applications with Quality of Service in Wi-Fi Networks.

112. On information and belief, one or more components of the Vtech Products and Services employs, provides, and dictates the performance of a method for comprising the step of sensing the communication medium (e.g., wireless medium) for an opportunity (e.g., transmission opportunity i.e., TXOP) to transmit message data units (e.g., MPDU, MMPDU, MSDU, etc.) without interference (e.g., collision avoidance method such as EDCA, etc.) from message data units (e.g., MPDU, MMPDU, MSDU, etc.) transmitted by a second station (e.g., another STA or AP in the WLAN network), according to sets of rules (e.g., transmission initiation rules) that vary by traffic classification (e.g., access category indicating different traffic types) yet are common to the first station (e.g., a transmitting STA or AP such as the accused product) and the second station (e.g., any other STA or AP in the network.

1.3 WMM Features

The features supported by WMM in this phase are as follows:

1. Capability negotiation independent of 802.11e. That is, WMM devices will not advertise 802.11e capability unless they also support those features independently. This is part of a forward compatibility strategy which is described in detail in a subsequent paragraph.
2. Frame formats and over the air protocols will be based on those currently proposed for 802.11e. However, no attempt will be made to track future changes in 802.11e and reflect them back into the WMM specification. Divergence between the two specifications is a necessary side effect of the need to freeze the WMM specification as soon as possible.
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4. Interfaces to the MAC which signal per-packet priority will be consistent with those used for Ethernet, both in terms of the driver API and bridging to other 802 link layers via an 802.1D bridge.
5. The number of exposed queues will be fixed at four. A fixed mapping of priority information carried in the 802.1D Priority field to those four queues will be defined, together with suggested uses for each priority consistent with the suggested uses in 802.1D.

2.2.2 WMM Parameter Element

The WMM Parameter Element contains a set of parameters (EDCA parameters) for the EDCF channel access protocol and is shown in Figure 8. The fields contained in the WMM Parameter Element are listed in Table 5. The WMM Parameter Element contains a QoS Info field. The format of the QoS Info field is shown in Figure 6 and Figure 7. The QoS Info field contains the Parameter Set Count, which is initially arbitrary and is incremented each time any of the AC parameters changes. The reserved bits are set to 0 upon transmission and shall be ignored upon reception.

Octets: 1	1	3	1	1	1	1	1	16
ElementID	Length	OUI	OUI Type	OUI Subtype	Version	QoS Info field	Reserved	AC Parameters

Figure 8 WMM Parameter Element

Table 5 WMM Parameter Element Field Values

Field	Value
Element ID	221
Length	24
OUI	`00:50:f2 (hex)`
OUI Type	2
OUI Subtype	1
Version	1
QoS Info field	See Figure 6 and Figure 7
Reserved	0
AC Parameters Best_Effort	AC Parameters Record AC_BE
AC Parameters Background	AC Parameters Record AC_BK
AC Parameters Video	AC Parameters Record AC_VI
AC Parameters Voice	AC Parameters Record AC_VO

Table 13 Default WMM Parameters

AC	CW_{min}	CW_{max}	AIFSN	TXOP Limit (802.11b)	TXOP Limit (802.11a/g)
AC_BK	aCW_{min}	aCW_{max}	7	0	0
AC_BE	aCW_{min}	aCW_{max}	3	0	0
AC_VI	$(aCW_{min} + 1)/2 - 1$	aCW_{min}	2	6.016ms	3.008ms
AC_VO	$(aCW_{min}+1)/4 - 1$	$(aCW_{min}+1)/2 - 1$	2	3.264ms	1.504ms

3.1.1 Procedure at an AP

An AP that supports WMM shall include either a WMM Information Element or a WMM Parameter Element in every beacon. In response to a probe request, a WMM-enabled AP shall include a WMM Parameter Element in its probe response.

3.1.2 Procedure at a STA in an Infrastructure Network

A WMM-enabled STA shall determine the WMM capability of an AP with which it wishes to associate before transmitting an association request to it. It may do this either passively, by receiving a beacon frame, or actively, by transmitting a probe request to it.

3.2.2 WMM Parameters in an Infrastructure Network

A WMM-enabled AP may arbitrarily determine values for the parameters CW_{min}, CW_{max}, AIFS and TXOP limit for each of the four access categories. An AP may change the values of these parameters at any beacon time.

An AP shall include a WMM Parameter Element containing its currently determined values for all WMM parameters in all beacons, beginning within two or more DTIM periods when WMM parameters have changed. The AC parameter set count that is contained in the WMM Parameter Element and WMM Information Element is incremented following the change of one or multiple WMM parameters. An AP shall include a WMM Parameter element in probe response frames and in association response frames.

The STA is required to monitor the change of WMM parameters conveyed in the WMM Parameter Element in beacons and shall update its values accordingly. Prior to association (and used only for transmitting probe, authentication and association request frames) it shall set these parameters to the default values shown in Table 13.

A STA shall subsequently update the values of these parameters from any successfully received probe response frame, association response frame, which are addressed to the STA or beacon frame transmitted by the AP with which the STA is associated. It is the responsibility of the STA to use the current WMM parameters for accessing the wireless medium.

AC (Access category): A label for the common set of enhanced distributed channel access (EDCA) parameters that are used by a WMM STA to contend for the channel in order to transmit MSDUs with certain priorities. WMM defines 4 ACs.

See e.g., screen captures from WMM Specification Version 1.1.

The collision resolution algorithm that is responsible for traffic prioritization is probabilistic and depends on two timing parameters that vary for each AC (Figure 4):

- the minimum interframe space, or Arbitrary Inter-Frame Space Number (AIFSN)
- the Contention Window (CW), sometimes referred to as the Random Backoff Wait.

After each collision the CW is doubled until a maximum value (also dependent on the AC) is reached. After successful transmission, the CW is reset to its initial, AC dependant value. The AC with the lowest backoff value gets the TXOP. As frames with the highest AC tend to have the lowest backoff values, they are more likely to get a TXOP.

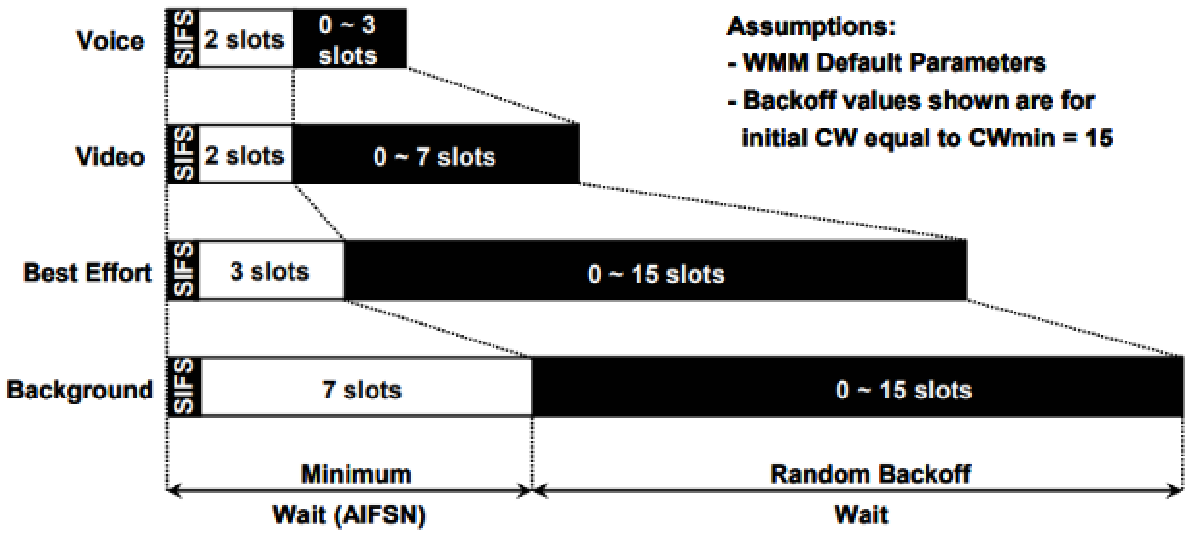

Figure 4. WMM AC Timing

Once a client gains a TXOP, it is allowed to transmit for a given time that depends on the AC and the PHY rate. For instance, the TXOP limit ranges from 0.2 ms (background

See e.g., screen captures from the Wi-Fi Certified for WMM – Support for Multimedia Applications with Quality of Service in Wi-Fi Networks.

TXOP (Transmission Opportunity): An interval of time when a particular WMM STA has the right to initiate transmissions onto the wireless medium (WM).

3.4.2 Transmit Opportunities & TXOP Limits

There are two modes of TXOP defined, EDCA TXOP and continuation TXOP.

An EDCA TXOP occurs when the EDCA rules permit access to the medium.

A continuation TXOP occurs when a channel access function retains the right to access the medium following the completion of a frame exchange sequence, such as on receipt of an Ack frame.

The TXOP limit duration values for each AC are advertised by the QAP in the WMM Parameter Element in Beacons, Probe response, and Association response frames. A TXOP limit value of 0 indicates that a single MSDU or MMPDU in addition to a possible RTS/CTS or CTS to itself may be transmitted at any PHY rate for each TXOP.

3.4.3 Obtaining an EDCA TXOP

Each channel access timer shall maintain a backoff function timer, which has a value measured in backoff slots.

The duration AIFS[AC] is a duration derived from the value AIFSN[AC] by the relation

$$\text{AIFS[AC]} = \text{AIFSN[AC]} \times \text{aSlotTime} + \text{aSIFSTime}$$

An EDCA TXOP is granted to a channel access function when the channel access function determines that it shall initiate the transmission of a frame exchange sequence. Transmission initiation shall be determined according to the following rules:

On specific slot boundaries, each channel access function shall make a determination to perform one and only one of the following functions:

a) Initiate the transmission of a frame exchange sequence for that access function

b) Decrement the backoff timer for that access function

c) Invoke the backoff procedure due to an internal collision

d) Do nothing for that access function.

The specific slot boundaries at which exactly one of these operations shall be performed are defined as follows, for each channel access function:

a) Following AIFSN[AC] * aSlotTime – aRxTxTurnaraoundTime of medium idle indication after SIFS (not necessarily idle time) after the last busy medium on the antenna, if the last busy medium indication was the result of a frame reception with a correct FCS ; or

b) Following EIFS – DIFS + AIFSN[AC] * aSlotTime of medium idle indication after the last indicated busy medium as determined by the carrier sense mechanism if the last busy medium indication was the result of a frame reception with an FCS error or PHY-RXEND.indication (RXERROR), where the value of RXERROR is not NoError.

c) When any other channel access function at this QSTA transmitted a frame requiring acknowledgement, the earlier of:

   a. the end of the ACK-Timeout interval timed from the PHY_TXEND.confirm, followed by AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE Time

   b. at the end of the first AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE medium after the PHY-RXEND.indication when a PHY-RXSTART.indication occurs as specified in subclause 9.2.8 of [1],

d) following AIFSN[AC] * aSlotTime– aTxRxTurnaraoundTime of medium idle indication after SIFS (not necessarily idle time) after the last indicated busy medium on the antenna that was the result of a transmission of a frame for any channel access function and which did not require an acknowledgement; or

e) following AIFSN[AC] * aSlotTimeof medium idle time indication after the last indicated idle medium as indicated by the carrier sense mechanism that is not covered by a) through d).

f) following aSlotTime of medium idle indication which occurs immediately after a decrement of the backoff counter for that channel access function.

At each of the above-described specific slot boundaries, each channel access function shall initiate a transmission sequence, if:

a) there is a frame available for transmission at that channel access function, and

b) the backoff timer for that channel access function has a value of zero, and

c) initiation of a transmission sequence is not allowed to commence at this time for a channel access function of higher UP.

At each of the above-described specific slot boundaries, each channel access function shall decrement the backoff timer by one, if:

a) The backoff timer for that channel access function has a value which is greater than zero.

At each of the above-described specific slot boundaries, each channel access function shall invoke the backoff procedure due to an internal collision, if:

a) There is a frame available for transmission at that channel access function, and

b) the backoff timer for that channel access function has a value of zero, and

c) initiation of a transmission sequence is allowed to commence at this time for a channel access function of higher UP.

At each of the above-described specific slot boundaries, each channel access function shall do nothing, if none of the above actions is taken.

See e.g., screen captures from WMM Specification Version 1.1.

113. On information and belief, one or more components of the VTech Products and Services employs, provides, and dictates the performance of a method for comprising the step of further including attempting to retransmit (e.g., retransmitting after failed transmission), after a respective interval (e.g., backoff timer) defined differently by each said set of rules (e.g., rules defined by channel access mechanism), any message data unit (e.g., MPDU, MMPDU, MSDU, etc.) transmitted over the communication medium (e.g., wireless medium) by a station (e.g., a WMM STA or AP such as the accused product) that collides with a message data unit (e.g., MPDU, MMPDU, MSDU, etc.) transmitted by another station (e.g., other STA with the WLAN network).

3.4.6 Retransmit Procedures

If a STA or AP, in an infrastructure BSS or an IBSS, transmits frames to a destination using QoS data types, it may following a failed transmission of a frame attempt to transmit another frame with a different access category to the same or any other destination. The STA has to contend for the medium when transmitting another frame with a different access category to the same or any other destination using the rules defined in [1].

If a STA or an AP does not use QoS data types when transmitting frames to a particular receiver address, once an initial attempt, excluding internal collisions, has been made to transmit a frame, it shall not transmit other data frames to that receiver address until the first frame is either successfully transmitted or discarded. It may, however, transmit other frames to different receiver addresses.

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Collisions between contending EDCAFs within a QSTA are resolved within the QSTA so that the data frames from the higher priority AC receives the TXOP and the data frames from the lower priority colliding AC(s) behave as if there were an external collision on the WM. Note, however, that this collision behavior does not include setting retry bits in the MAC headers of MPDUs at the head of the lower priority ACs, as would be done after a transmission attempt that was unsuccessful due to an actual external collision on the WM.

See e.g., screen captures from IEEE 802-11e Standard.

Each channel access function shall maintain a state variable CW[AC], which shall be initialized to the value of the parameter CWmin[AC] (see section 3.2).

If a frame is successfully transmitted for a specific channel access function, indicated by either the successful reception of a CTS in response to an RTS, or the successful reception of an Ack in response to a unicast MPDU, or by transmitting a multicast frame or a frame with "no acknowledgement" policy, CW[AC] shall be reset to CWmin[AC].

The backoff procedure shall be invoked for a channel access function when either:

a) A frame with that AC is requested to be transmitted and the medium is busy as indicated by either physical or virtual carrier sense, and the backoff timer has a value of zero for that AC.

b) The final transmission by the TXOP holder initiated during the TXOP for that AC was successful

c) The transmission of a frame of that AC fails, indicated by a failure to receive a CTS in response to an RTS, or a failure to receive an Ack that was expected in response to a unicast MPDU or MMPDU.

d) The transmission attempt collides internally with another channel access function of an AC that has higher priority, that is, two or more channel access functions in the same STA or AP are granted a TXOP at the same time.

If the backoff procedure is invoked because of a failure event (either reason c) or d) above) the value of CW[AC] shall be updated as follows before invoking the backoff procedure:

a) if the short or long retry count for the STA has reached aShortRetryLimit or aLongRetryLimit[AC] respectively, CW[AC] will be reset to CWmin[AC].

b) Otherwise,

1) if CW[AC] is less than CWmax[AC], CW[AC] shall be set to the value (CW[AC]+1)*2-1
2) if CW[AC] is equal to CWmax[AC], CW[AC] shall remain unchanged for the remainder of any retries

Following the update of the value of CW[AC], the backoff timer is set to an integer value chosen randomly with a uniform distribution taking values in the range (0,CW[AC]) inclusive.

See e.g., screen captures from WMM Specification Version 1.1.

WMM prioritization works as shown in Figure 3 and Figure 4. Applications assign each data packet to a given AC (Figure 3). Packets are then added to one of four independent transmit queues (one per AC; i.e., voice, video, best effort, or background) in the client. The client has an internal collision resolution mechanism to address collision among different queues, which selects the frames with the highest priority to transmit. The same mechanism deals with external collision, to determine which client should be granted the Opportunity to Transmit (TXOP).

The collision resolution algorithm that is responsible for traffic prioritization is probabilistic and depends on two timing parameters that vary for each AC (Figure 4):

- the minimum interframe space, or Arbitrary Inter-Frame Space Number (AIFSN)
- the Contention Window (CW), sometimes referred to as the Random Backoff Wait.

After each collision the CW is doubled until a maximum value (also dependent on the AC) is reached. After successful transmission, the CW is reset to its initial, AC dependant value. The AC with the lowest backoff value gets the TXOP. As frames with the highest AC tend to have the lowest backoff values, they are more likely to get a TXOP.

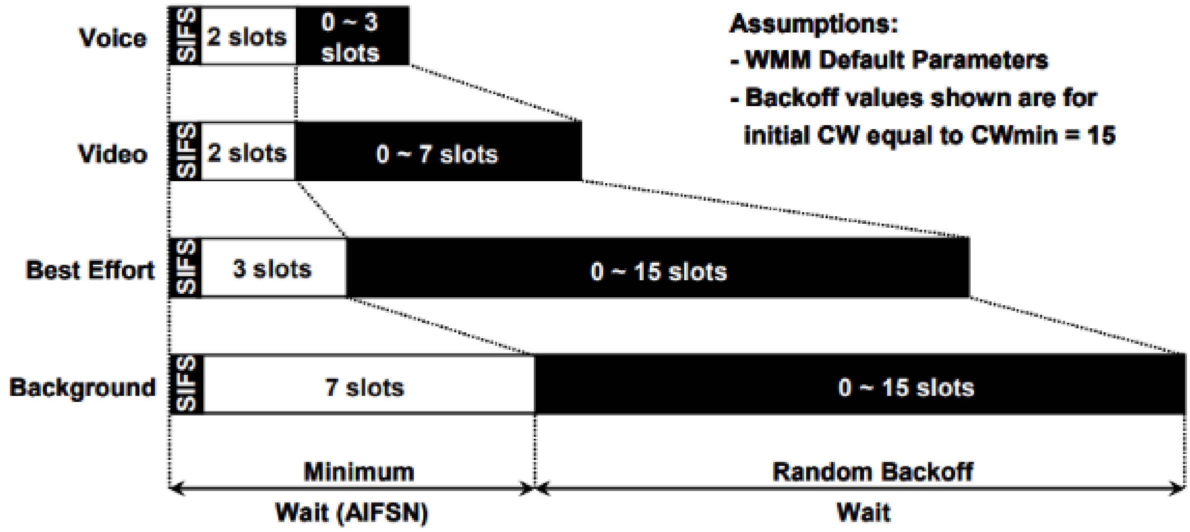

Figure 4. WMM AC Timing

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See e.g., screen captures from the Wi-Fi Certified for WMM – Support for Multimedia Applications with Quality of Service in Wi-Fi Networks.

A STA desiring to initiate transfer of data MPDUs and/or MMPDUs shall invoke the CS mechanism (see 9.2.1) to determine the busy/idle state of the medium. If the medium is busy, the STA shall defer until the medium is determined to be idle without interruption for a period of time equal to DIFS when the last frame detected on the medium was received correctly, or after the medium is determined to be idle without interruption for a period of time equal to EIFS when the last frame detected on the medium was not received correctly. After this DIFS or EIFS medium idle time, the STA shall then generate a random backoff period for an additional deferral time before transmitting, unless the backoff timer already contains a nonzero value, in which case the selection of a random number is not needed and not performed. This process minimizes collisions during contention between multiple STAs that have been deferring to the same event.

Backoff Time = Random() × aSlotTime

where

Random() = Pseudo-random integer drawn from a uniform distribution over the interval [0,CW], where CW is an integer within the range of values of the PHY characteristics aCWmin and aCWmax, aCWmin ≤ CW ≤ aCWmax. It is important that designers recognize the need for statistical independence among the random number streams among STAs.

aSlotTime = The value of the correspondingly named PHY characteristic.

See e.g., screen capture from the IEEE 802-11 Standard.

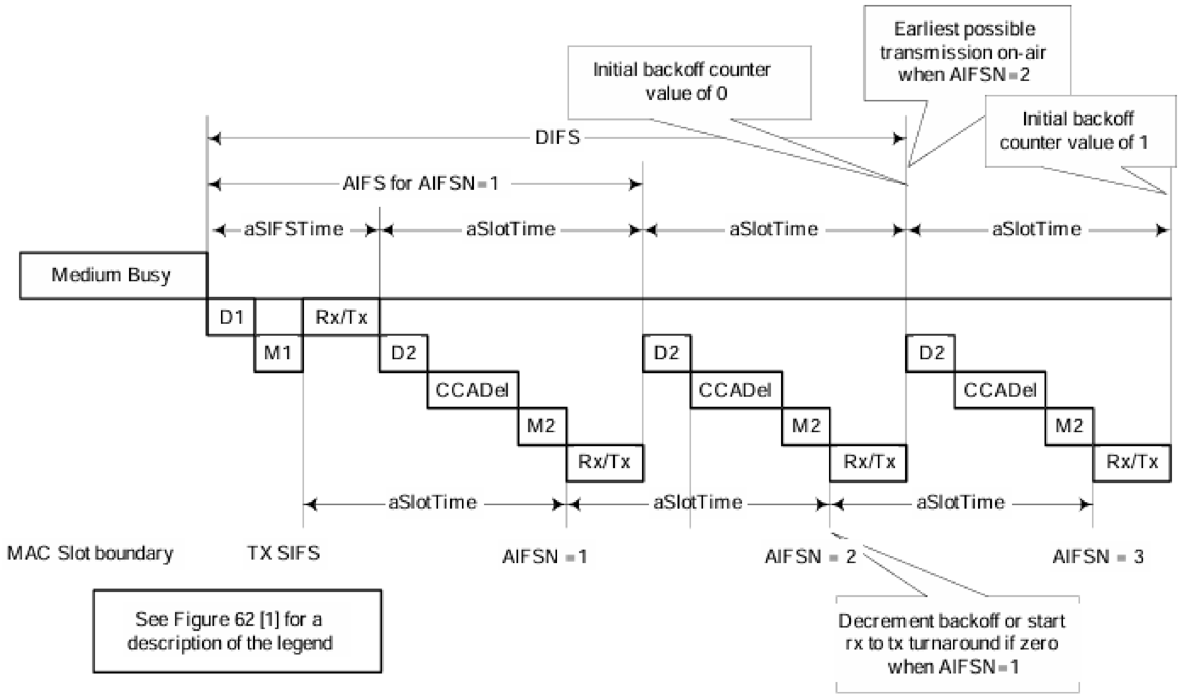

Figure 18 EDCA Mechanism Timing

The specific slot boundaries at which exactly one of these operations shall be performed are defined as follows, for each channel access function:

a) Following AIFSN[AC] * aSlotTime – aRxTxTurnaraoundTime of medium idle indication after SIFS (not necessarily idle time) after the last busy medium on the antenna, if the last busy medium indication was the result of a frame reception with a correct FCS ; or

b) Following EIFS – DIFS + AIFSN[AC] * aSlotTime of medium idle indication after the last indicated busy medium as determined by the carrier sense mechanism if the last busy medium indication was the result of a frame reception with an FCS error or PHY-RXEND.indication (RXERROR), where the value of RXERROR is not NoError.

c) When any other channel access function at this QSTA transmitted a frame requiring acknowledgement, the earlier of:

   a. the end of the ACK-Timeout interval timed from the PHY_TXEND.confirm, followed by AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE Time

   b. at the end of the first AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE medium after the PHY-RXEND.indication when a PHY-RXSTART.indication occurs as specified in subclause 9.2.8 of [1],

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e) following AIFSN[AC] * aSlotTimeof medium idle time indication after the last indicated idle medium as indicated by the carrier sense mechanism that is not covered by a) through d).

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At each of the above-described specific slot boundaries, each channel access function shall initiate a transmission sequence, if:

a) there is a frame available for transmission at that channel access function, and

b) the backoff timer for that channel access function has a value of zero, and

c) initiation of a transmission sequence is not allowed to commence at this time for a channel access function of higher UP.

At each of the above-described specific slot boundaries, each channel access function shall decrement the backoff timer by one, if:

a) The backoff timer for that channel access function has a value which is greater than zero.

At each of the above-described specific slot boundaries, each channel access function shall invoke the backoff procedure due to an internal collision, if:

a) There is a frame available for transmission at that channel access function, and

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3.4.6 Retransmit Procedures

If a STA or AP, in an infrastructure BSS or an IBSS, transmits frames to a destination using QoS data types, it may following a failed transmission of a frame attempt to transmit another frame with a different access category to the same or any other destination. The STA has to contend for the medium when transmitting another frame with a different access category to the same or any other destination using the rules defined in [1].

If a STA or an AP does not use QoS data types when transmitting frames to a particular receiver address, once an initial attempt, excluding internal collisions, has been made to transmit a frame, it shall not transmit other data frames to that receiver address until the first frame is either successfully transmitted or discarded. It may, however, transmit other frames to different receiver addresses.

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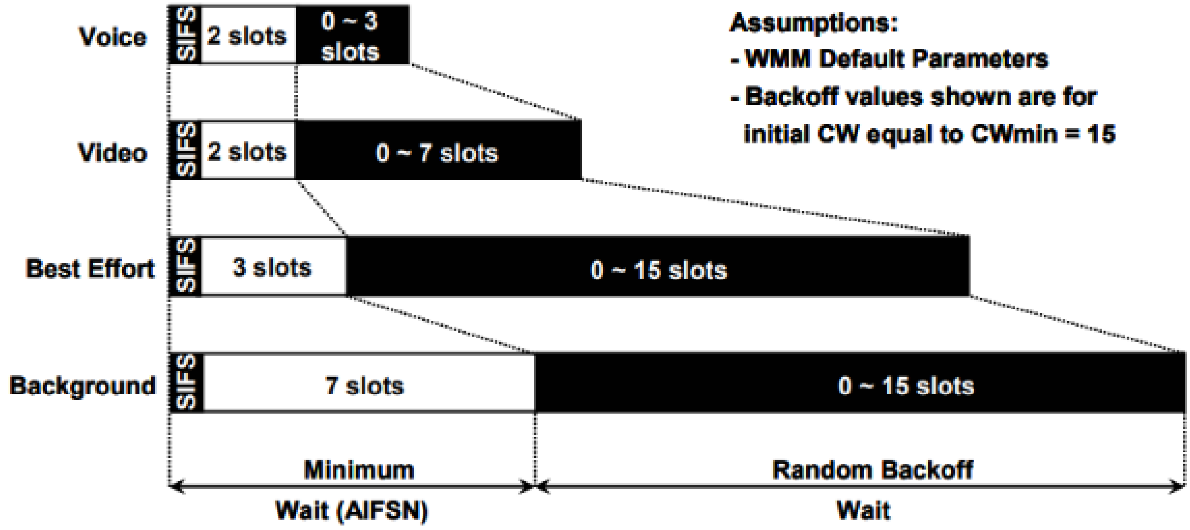

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$$\text{Backoff Time} = \text{Random()} \times \text{aSlotTime}$$

where

Random() = Pseudo-random integer drawn from a uniform distribution over the interval [0,CW], where CW is an integer within the range of values of the PHY characteristics aCWmin and aCWmax, aCWmin $\leq$ CW $\leq$ aCWmax. It is important that designers recognize the need for statistical independence among the random number streams among STAs.

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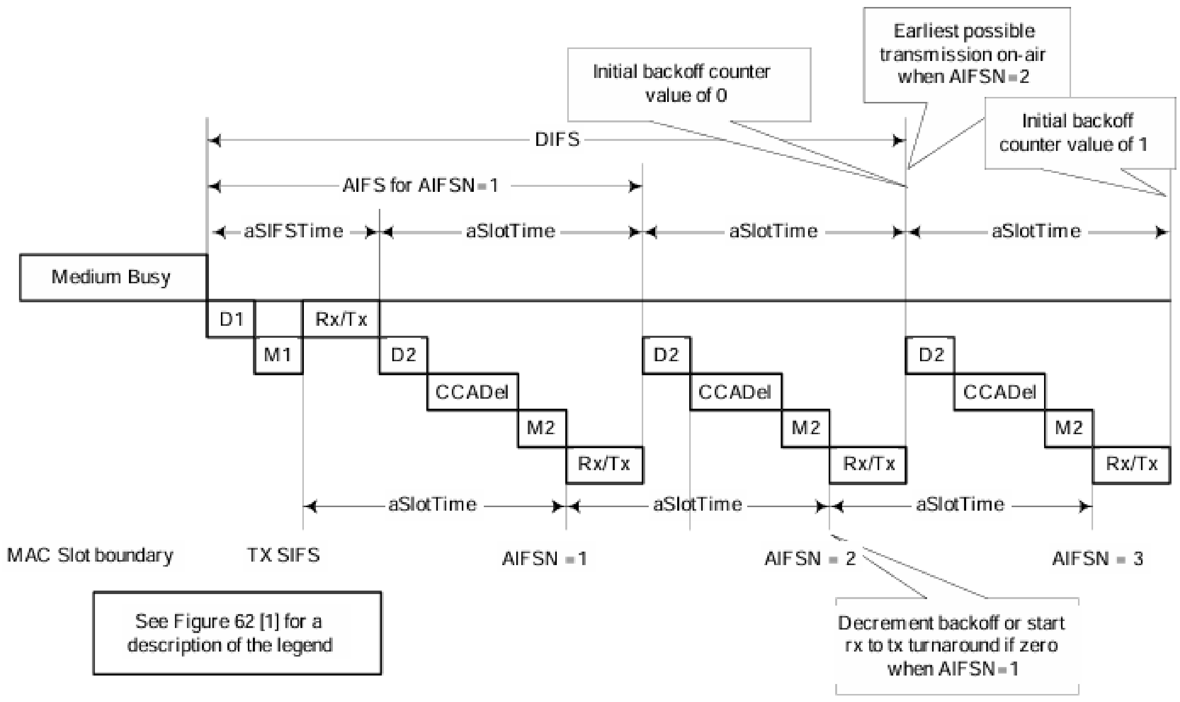

Figure 18 EDCA Mechanism Timing

The specific slot boundaries at which exactly one of these operations shall be performed are defined as follows, for each channel access function:

a) Following AIFSN[AC] * aSlotTime – aRxTxTurnaraoundTime of medium idle indication after SIFS (not necessarily idle time) after the last busy medium on the antenna, if the last busy medium indication was the result of a frame reception with a correct FCS ; or

b) Following EIFS – DIFS + AIFSN[AC] * aSlotTime of medium idle indication after the last indicated busy medium as determined by the carrier sense mechanism if the last busy medium indication was the result of a frame reception with an FCS error or PHY-RXEND.indication (RXERROR), where the value of RXERROR is not NoError.

c) When any other channel access function at this QSTA transmitted a frame requiring acknowledgement, the earlier of:

   a. the end of the ACK-Timeout interval timed from the PHY_TXEND.confirm, followed by AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE Time

   b. at the end of the first AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE medium after the PHY-RXEND.indication when a PHY-RXSTART.indication occurs as specified in subclause 9.2.8 of [1],

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At each of the above-described specific slot boundaries, each channel access function shall initiate a transmission sequence, if:

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3.4.6 Retransmit Procedures

If a STA or AP, in an infrastructure BSS or an IBSS, transmits frames to a destination using QoS data types, it may following a failed transmission of a frame attempt to transmit another frame with a different access category to the same or any other destination. The STA has to contend for the medium when transmitting another frame with a different access category to the same or any other destination using the rules defined in [1].

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The backoff procedure shall be invoked for a channel access function when either:

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The collision resolution algorithm that is responsible for traffic prioritization is probabilistic and depends on two timing parameters that vary for each AC (Figure 4):

- the minimum interframe space, or Arbitrary Inter-Frame Space Number (AIFSN)
- the Contention Window (CW), sometimes referred to as the Random Backoff Wait.

After each collision the CW is doubled until a maximum value (also dependent on the AC) is reached. After successful transmission, the CW is reset to its initial, AC dependant value. The AC with the lowest backoff value gets the TXOP. As frames with the highest AC tend to have the lowest backoff values, they are more likely to get a TXOP.

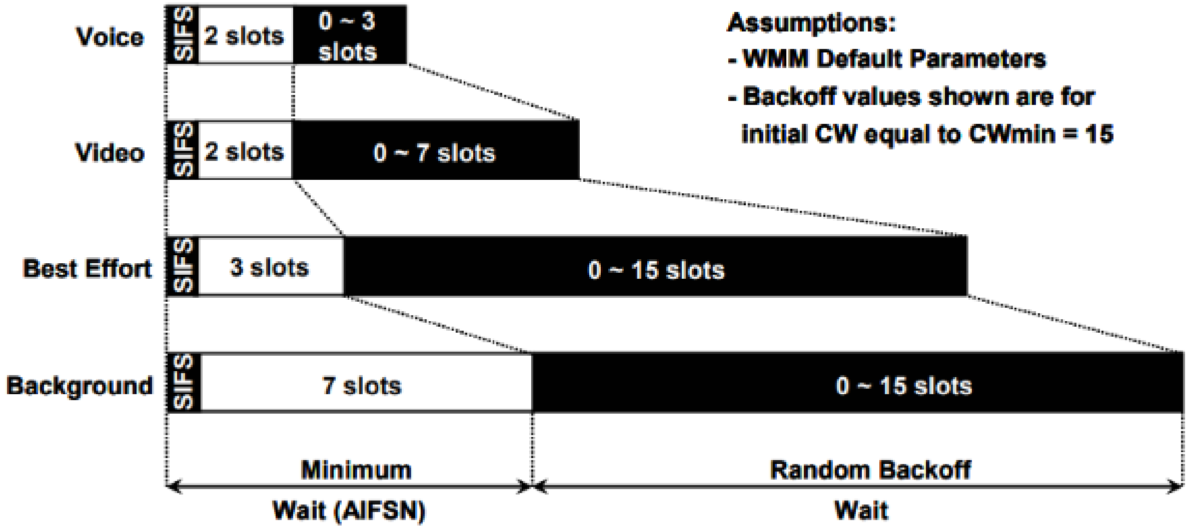

Figure 4. WMM AC Timing

Once a client gains a TXOP, it is allowed to transmit for a given time that depends on the AC and the PHY rate. For instance, the TXOP limit ranges from 0.2 ms (background

See e.g., screen captures from the Wi-Fi Certified for WMM – Support for Multimedia Applications with Quality of Service in Wi-Fi Networks.

A STA desiring to initiate transfer of data MPDUs and/or MMPDUs shall invoke the CS mechanism (see 9.2.1) to determine the busy/idle state of the medium. If the medium is busy, the STA shall defer until the medium is determined to be idle without interruption for a period of time equal to DIFS when the last frame detected on the medium was received correctly, or after the medium is determined to be idle without interruption for a period of time equal to EIFS when the last frame detected on the medium was not received correctly. After this DIFS or EIFS medium idle time, the STA shall then generate a random backoff period for an additional deferral time before transmitting, unless the backoff timer already contains a nonzero value, in which case the selection of a random number is not needed and not performed. This process minimizes collisions during contention between multiple STAs that have been deferring to the same event.

Backoff Time = Random() × aSlotTime

where

Random() = Pseudo-random integer drawn from a uniform distribution over the interval [0,CW], where CW is an integer within the range of values of the PHY characteristics aCWmin and aCW-max, aCWmin ≤ CW ≤ aCWmax. It is important that designers recognize the need for statistical independence among the random number streams among STAs.

aSlotTime = The value of the correspondingly named PHY characteristic.

See e.g., screen capture from the IEEE 802-11 Standard.

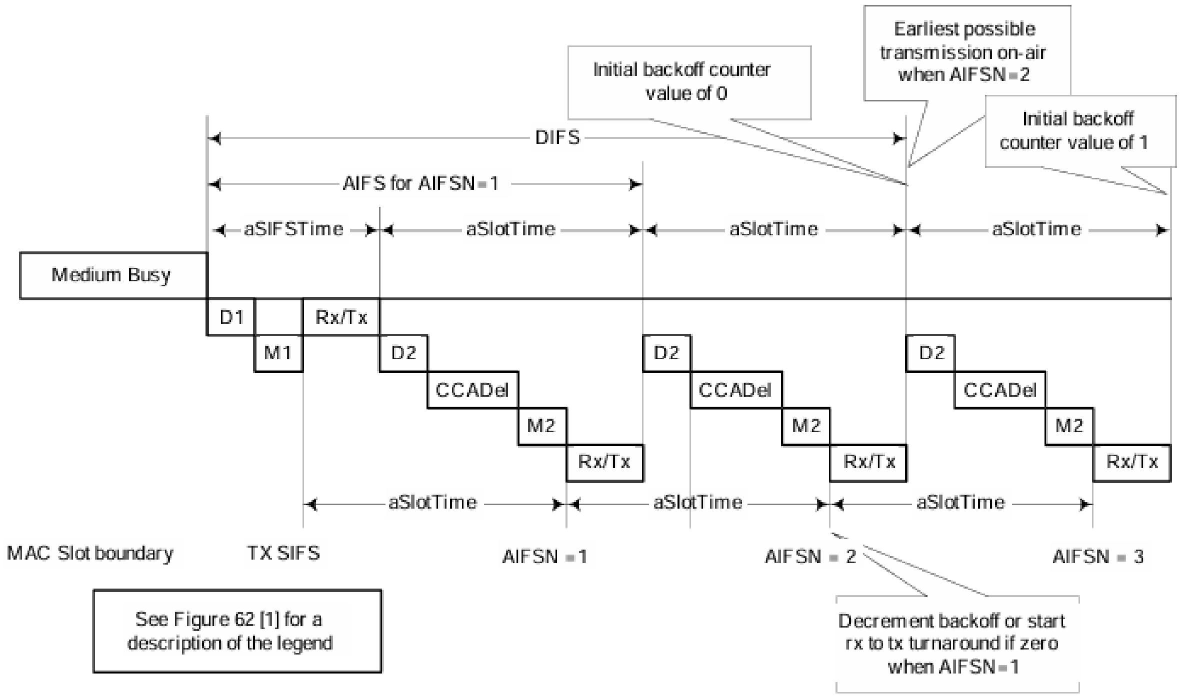

Figure 18 EDCA Mechanism Timing

The specific slot boundaries at which exactly one of these operations shall be performed are defined as follows, for each channel access function:

a) Following AIFSN[AC] * aSlotTime – aRxTxTurnaraoundTime of medium idle indication after SIFS (not necessarily idle time) after the last busy medium on the antenna, if the last busy medium indication was the result of a frame reception with a correct FCS ; or

b) Following EIFS – DIFS + AIFSN[AC] * aSlotTime of medium idle indication after the last indicated busy medium as determined by the carrier sense mechanism if the last busy medium indication was the result of a frame reception with an FCS error or PHY-RXEND.indication (RXERROR), where the value of RXERROR is not NoError.

c) When any other channel access function at this QSTA transmitted a frame requiring acknowledgement, the earlier of:

    a. the end of the ACK-Timeout interval timed from the PHY_TXEND.confirm, followed by AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE Time

    b. at the end of the first AIFSN[AC] * aSlotTime – aTxRxTurnaraoundTime of IDLE medium after the PHY-RXEND.indication when a PHY-RXSTART.indication occurs as specified in subclause 9.2.8 of [1],

d) following AIFSN[AC] * aSlotTime– aTxRxTurnaraoundTime of medium idle indication after SIFS (not necessarily idle time) after the last indicated busy medium on the antenna that was the result of a transmission of a frame for any channel access function and which did not require an acknowledgement; or

e) following AIFSN[AC] * aSlotTimeof medium idle time indication after the last indicated idle medium as indicated by the carrier sense mechanism that is not covered by a) through d).

f) following aSlotTime of medium idle indication which occurs immediately after a decrement of the backoff counter for that channel access function.

At each of the above-described specific slot boundaries, each channel access function shall initiate a transmission sequence, if:

a) there is a frame available for transmission at that channel access function, and

b) the backoff timer for that channel access function has a value of zero, and

c) initiation of a transmission sequence is not allowed to commence at this time for a channel access function of higher UP.

At each of the above-described specific slot boundaries, each channel access function shall decrement the backoff timer by one, if:

a) The backoff timer for that channel access function has a value which is greater than zero.

At each of the above-described specific slot boundaries, each channel access function shall invoke the backoff procedure due to an internal collision, if:

a) There is a frame available for transmission at that channel access function, and

b) the backoff timer for that channel access function has a value of zero, and

c) initiation of a transmission sequence is allowed to commence at this time for a channel access function of higher UP.

At each of the above-described specific slot boundaries, each channel access function shall do nothing, if none of the above actions is taken.

See e.g., screen captures from WMM Specification Version 1.1.

116. ThinkLogix has been damaged by and has suffered irreparable harm as a result of Vtech's infringement.

JURY DEMANDED

117. Pursuant to Federal Rule of Civil Procedure 38(b), ThinkLogix hereby requests a trial by jury on all issues so triable.

PRAYER FOR RELIEF

118. ThinkLogix respectfully requests this Court to enter judgment in ThinkLogix's favor and against VTech as follows:

a. finding that VTech has infringed one or more claims of the '373 patent under 35 U.S.C. § 271(a);

b. finding that VTech has infringed one or more claims of the '835 patent under 35 U.S.C. § 271(a) and (b);

c. finding that VTech has infringed one or more claims of the ’994 patent under 35 U.S.C. § 271(a);

d. finding that VTech has infringed one or more claims of the ’573 patent under 35 U.S.C. § 271(a);

e. finding that VTech has infringed one or more claims of the ’524 patent under 35 U.S.C. § 271(a);

f. finding that VTech has infringed one or more claims of the ’392 patent under 35 U.S.C. § 271(a);

g. awarding ThinkLogix damages under 35 U.S.C. § 284, or otherwise permitted by law;

h. awarding ThinkLogix pre-judgment and post-judgment interest on the damages award and costs;

i. declaring that VTech has willfully infringed one or more claims of the Patents-in-Suit;

j. awarding treble damages pursuant to 35 U.S.C. § 284 as a result of VTech’s willful conduct in relation to one or more claims of the Patents-in-Suit;

k. awarding cost of this action (including all disbursements) and attorney fees pursuant to 35 U.S.C. § 285, or as otherwise permitted by the law; and

l. awarding such other costs and further relief that the Court determines to be just and equitable.

Dated: April 22, 2024

Respectfully submitted,

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