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Chief Scientist & Co-founder

Recent Advances in Multipoint Video Server Architectures:

Scalability, Simulcasting, and SFUs

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history of multipoint video

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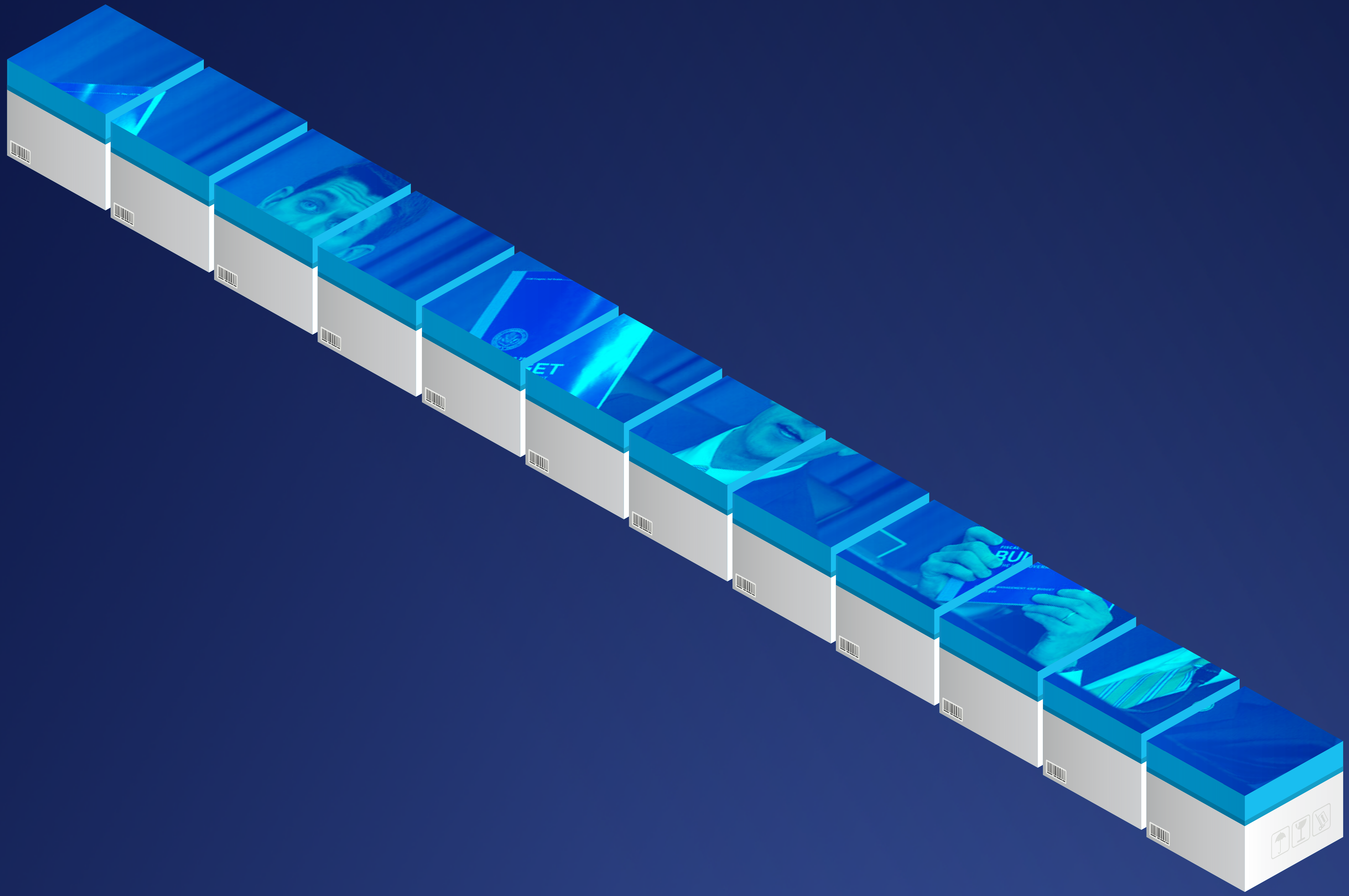

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name of stream

timeline

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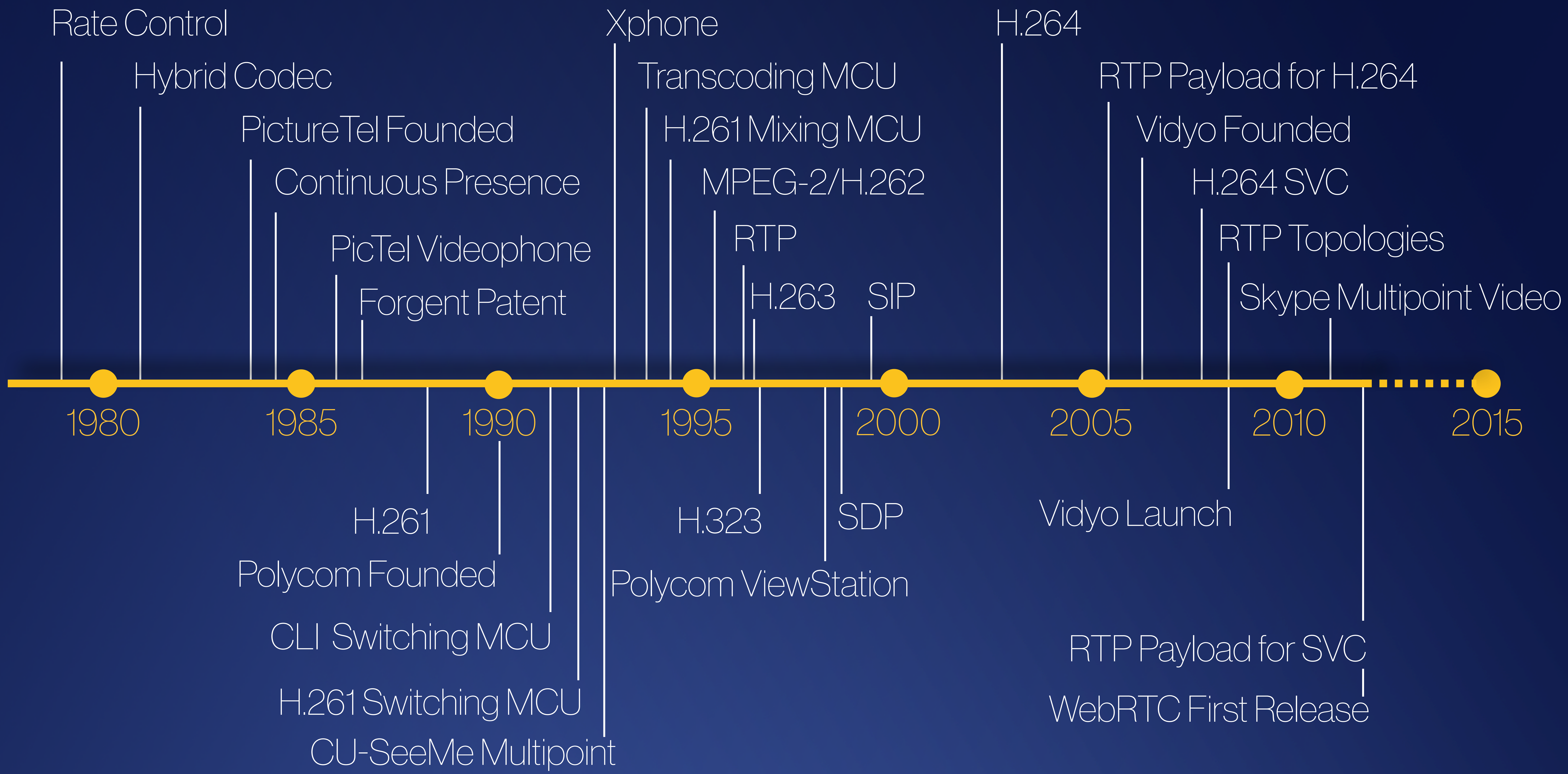

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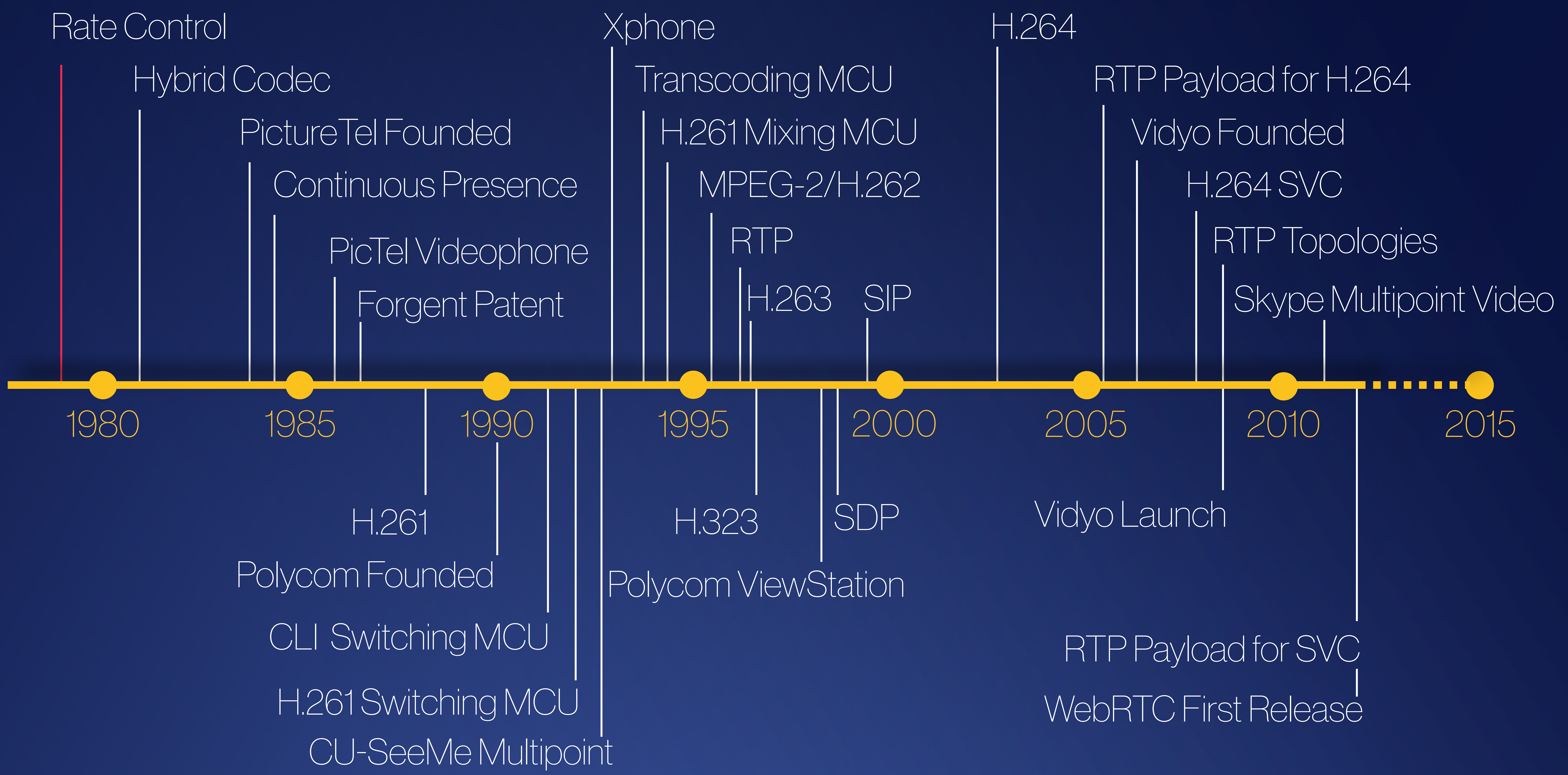

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Rate Control

December 1978
US Pat. Nr. 4,302,775
Compression Labs, Inc.

First CLI video patent,
rate control through
adaptive quantization

United States Patent [19] [11] 4,302,775

Widergren et al. [45] Nov. 24, 1981

[54] DIGITAL VIDEO COMPRESSION SYSTEM AND METHODS UTILIZING SCENE ADAPTIVE CODING WITH RATE BUFFER FEEDBACK

[75] Inventors: Robert D. Widergren, Saratoga; Wen-Hsiung Chen, Sunnyvale; Stanley C. Fralick, Saratoga; Andrew G. Tescher, Claremont, all of Calif.

[73] Assignee: Compression Labs, Inc., San Jose, Calif.

[21] Appl. No.: 969,991

[22] Filed: Dec. 15, 1978

[51] Int. Cl.³ H04N 7/12; H04N 9/32; G06F 15/20; G08C 9/00

[52] U.S. Cl. .... 358/136; 358/13; 340/347 DD; 364/514; 364/515; 364/582

[58] Field of Search .... 364/514, 515, 576, 582; 358/12, 13, 133, 138, 260, 261; 340/347 DD

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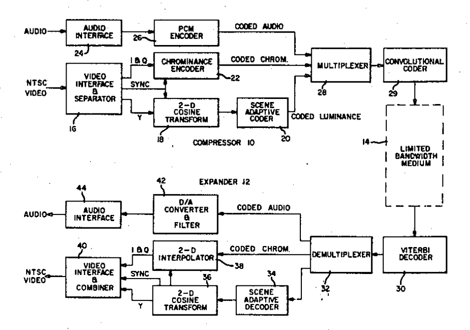
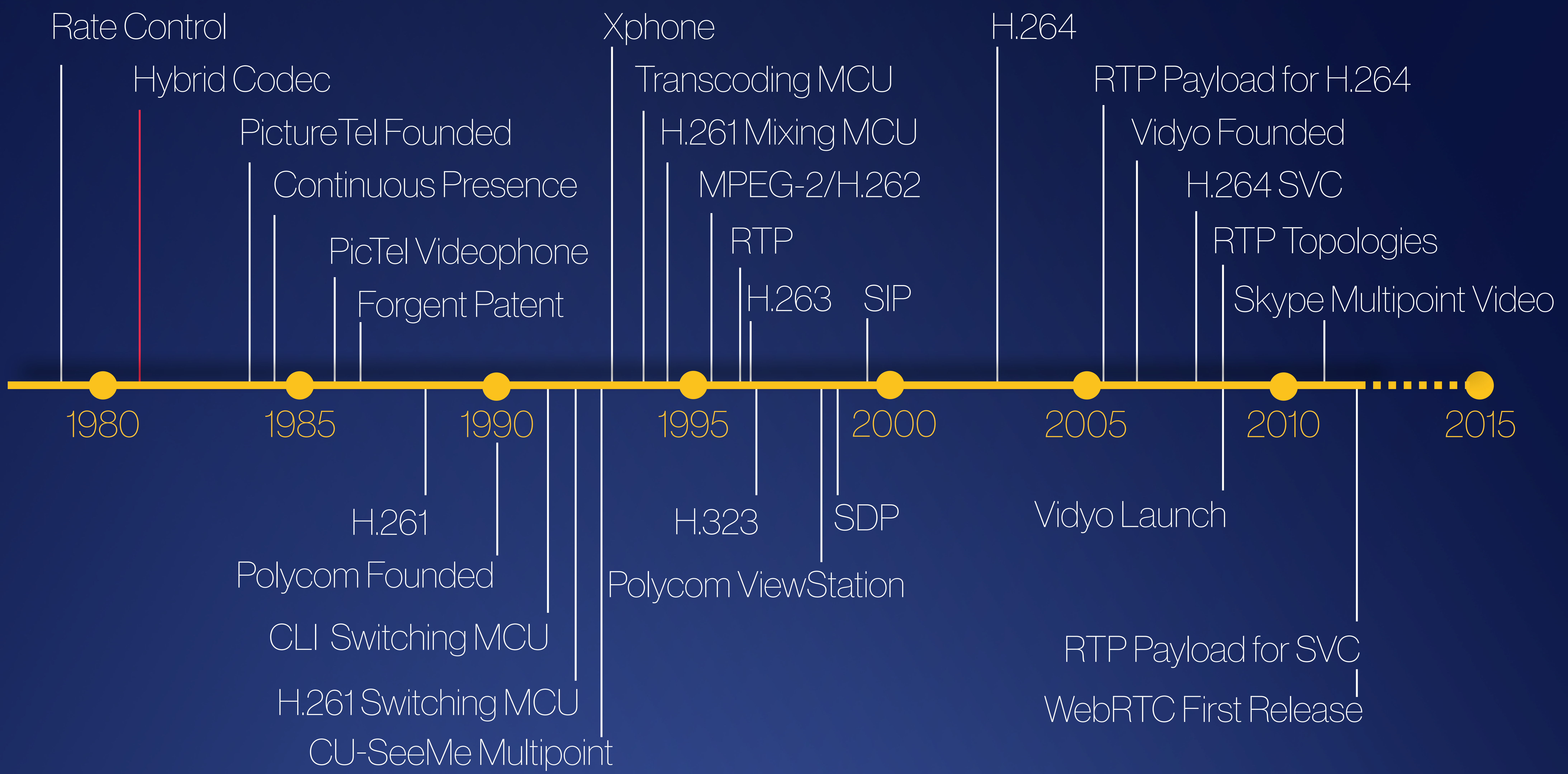
Primary Examiner—James W. Moffitt
Assistant Examiner—Aristotelis M. Psitos
Attorney, Agent, or Firm—David B. Harrison

[57] ABSTRACT

A digital video compression system and its methods for compressing digitalized video signals in real time at rates up to NTSC color broadcast rates are disclosed. The system compressor receives digitalized video frames divided into subframes, performs in a single pass a spatial domain to transform domain transformation in two dimensions of the picture elements of each subframe, normalizes the resultant coefficients by a normalization factor having a predetermined compression ratio component and an adaptive rate buffer capacity control feedback component, to provide compression, encodes the coefficients and stores them in a first rate buffer memory asynchronously at a high data transfer rate from which they are put out at a slower, synchronous rate. The compressor adaptively determines the rate buffer capacity control feedback component in relation to instantaneous data content of the rate buffer memory in relation to its capacity, and it controls the absolute quantity of data resulting from the normalization step so that the buffer memory is never completely emptied and never completely filled. In expansion, the system essentially mirrors the steps performed during compression. An efficient, high speed decoder forms an important aspect of the present invention. The compression system forms an important element of a disclosed color broadcast compression system.

7 Claims, 30 Drawing Figures





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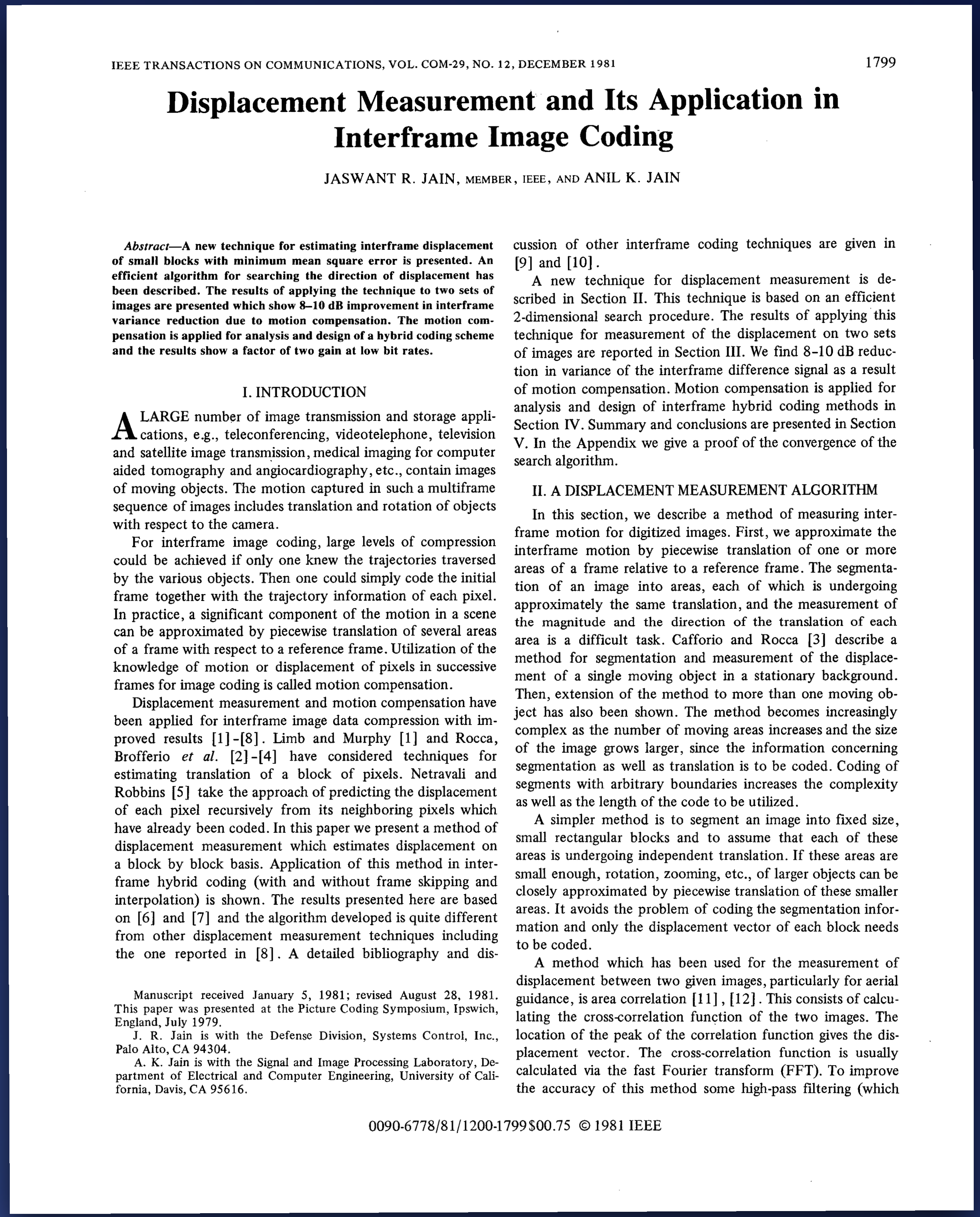
Hybrid Codec

December 1981

Jain & Jain

IEEE Trans. on Comm.

Block-based motion estimation with transform coding



IEEE TRANSACTIONS ON COMMUNICATIONS, VOL. COM-29, NO. 12, DECEMBER 1981 1799

Displacement Measurement and Its Application in Interframe Image Coding

JASWANT R. JAIN, MEMBER, IEEE, AND ANIL K. JAIN

Abstract—**A new technique for estimating interframe displacement of small blocks with minimum mean square error is presented. An efficient algorithm for searching the direction of displacement has been described. The results of applying the technique to two sets of images are presented which show 8–10 dB improvement in interframe variance reduction due to motion compensation. The motion compensation is applied for analysis and design of a hybrid coding scheme and the results show a factor of two gain at low bit rates.**

I. INTRODUCTION

A LARGE number of image transmission and storage applications, e.g., teleconferencing, videotelephone, television and satellite image transmission, medical imaging for computer aided tomography and angiocardiography, etc., contain images of moving objects. The motion captured in such a multiframe sequence of images includes translation and rotation of objects with respect to the camera.

For interframe image coding, large levels of compression could be achieved if only one knew the trajectories traversed by the various objects. Then one could simply code the initial frame together with the trajectory information of each pixel. In practice, a significant component of the motion in a scene can be approximated by piecewise translation of several areas of a frame with respect to a reference frame. Utilization of the knowledge of motion or displacement of pixels in successive frames for image coding is called motion compensation.

Displacement measurement and motion compensation have been applied for interframe image data compression with improved results [1]-[8]. Limb and Murphy [1] and Rocca, Brofferio *et al.* [2]-[4] have considered techniques for estimating translation of a block of pixels. Netravali and Robbins [5] take the approach of predicting the displacement of each pixel recursively from its neighboring pixels which have already been coded. In this paper we present a method of displacement measurement which estimates displacement on a block by block basis. Application of this method in interframe hybrid coding (with and without frame skipping and interpolation) is shown. The results presented here are based on [6] and [7] and the algorithm developed is quite different from other displacement measurement techniques including the one reported in [8]. A detailed bibliography and discussion of other interframe coding techniques are given in [9] and [10].

A new technique for displacement measurement is described in Section II. This technique is based on an efficient 2-dimensional search procedure. The results of applying this technique for measurement of the displacement on two sets of images are reported in Section III. We find 8-10 dB reduction in variance of the interframe difference signal as a result of motion compensation. Motion compensation is applied for analysis and design of interframe hybrid coding methods in Section IV. Summary and conclusions are presented in Section V. In the Appendix we give a proof of the convergence of the search algorithm.

II. A DISPLACEMENT MEASUREMENT ALGORITHM

In this section, we describe a method of measuring interframe motion for digitized images. First, we approximate the interframe motion by piecewise translation of one or more areas of a frame relative to a reference frame. The segmentation of an image into areas, each of which is undergoing approximately the same translation, and the measurement of the magnitude and the direction of the translation of each area is a difficult task. Cafforio and Rocca [3] describe a method for segmentation and measurement of the displacement of a single moving object in a stationary background. Then, extension of the method to more than one moving object has also been shown. The method becomes increasingly complex as the number of moving areas increases and the size of the image grows larger, since the information concerning segmentation as well as translation is to be coded. Coding of segments with arbitrary boundaries increases the complexity as well as the length of the code to be utilized.

A simpler method is to segment an image into fixed size, small rectangular blocks and to assume that each of these areas is undergoing independent translation. If these areas are small enough, rotation, zooming, etc., of larger objects can be closely approximated by piecewise translation of these smaller areas. It avoids the problem of coding the segmentation information and only the displacement vector of each block needs to be coded.

A method which has been used for the measurement of displacement between two given images, particularly for aerial guidance, is area correlation [11], [12]. This consists of calculating the cross-correlation function of the two images. The location of the peak of the correlation function gives the displacement vector. The cross-correlation function is usually calculated via the fast Fourier transform (FFT). To improve the accuracy of this method some high-pass filtering (which

Manuscript received January 5, 1981; revised August 28, 1981. This paper was presented at the Picture Coding Symposium, Ipswich, England, July 1979.

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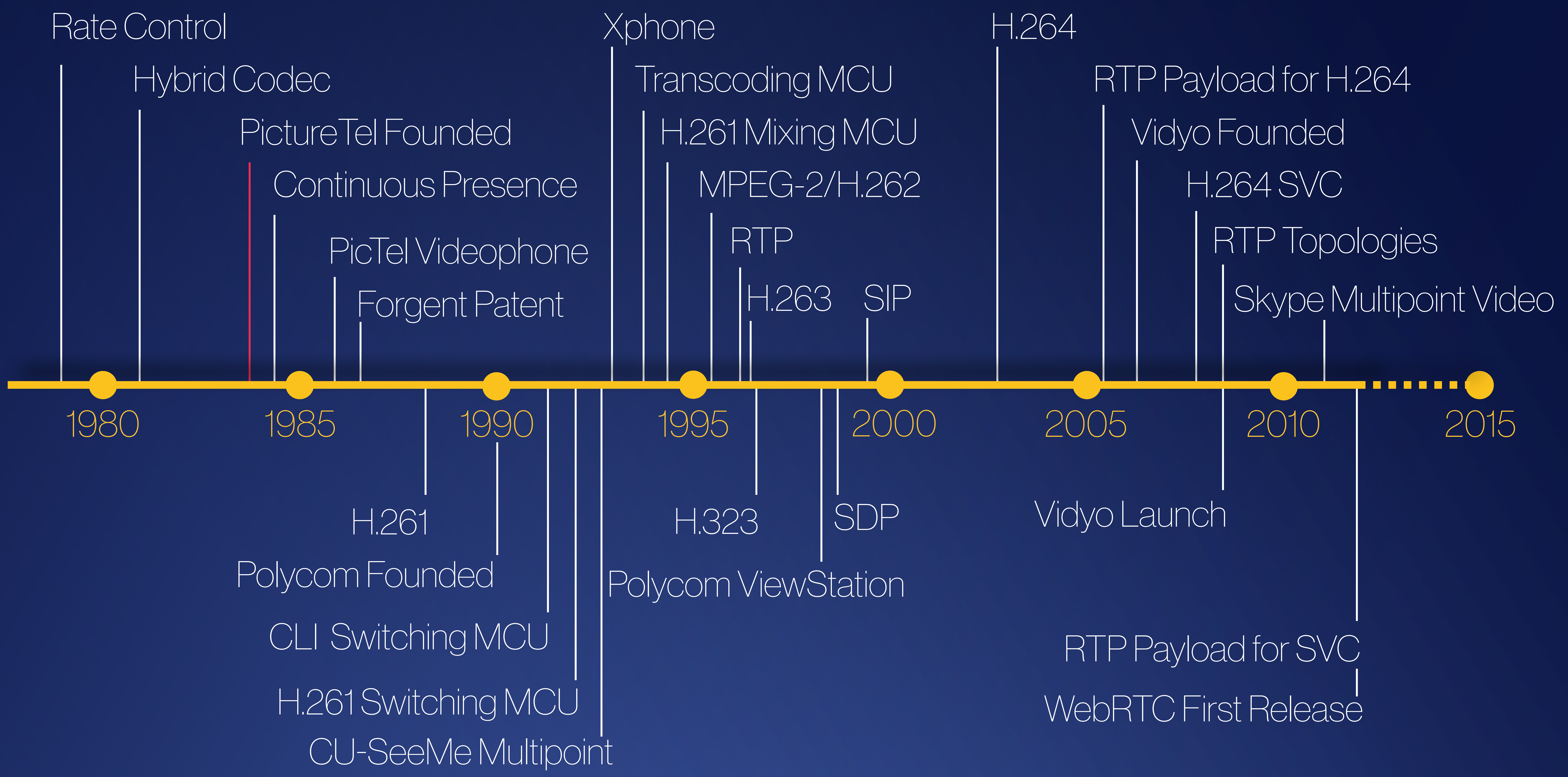

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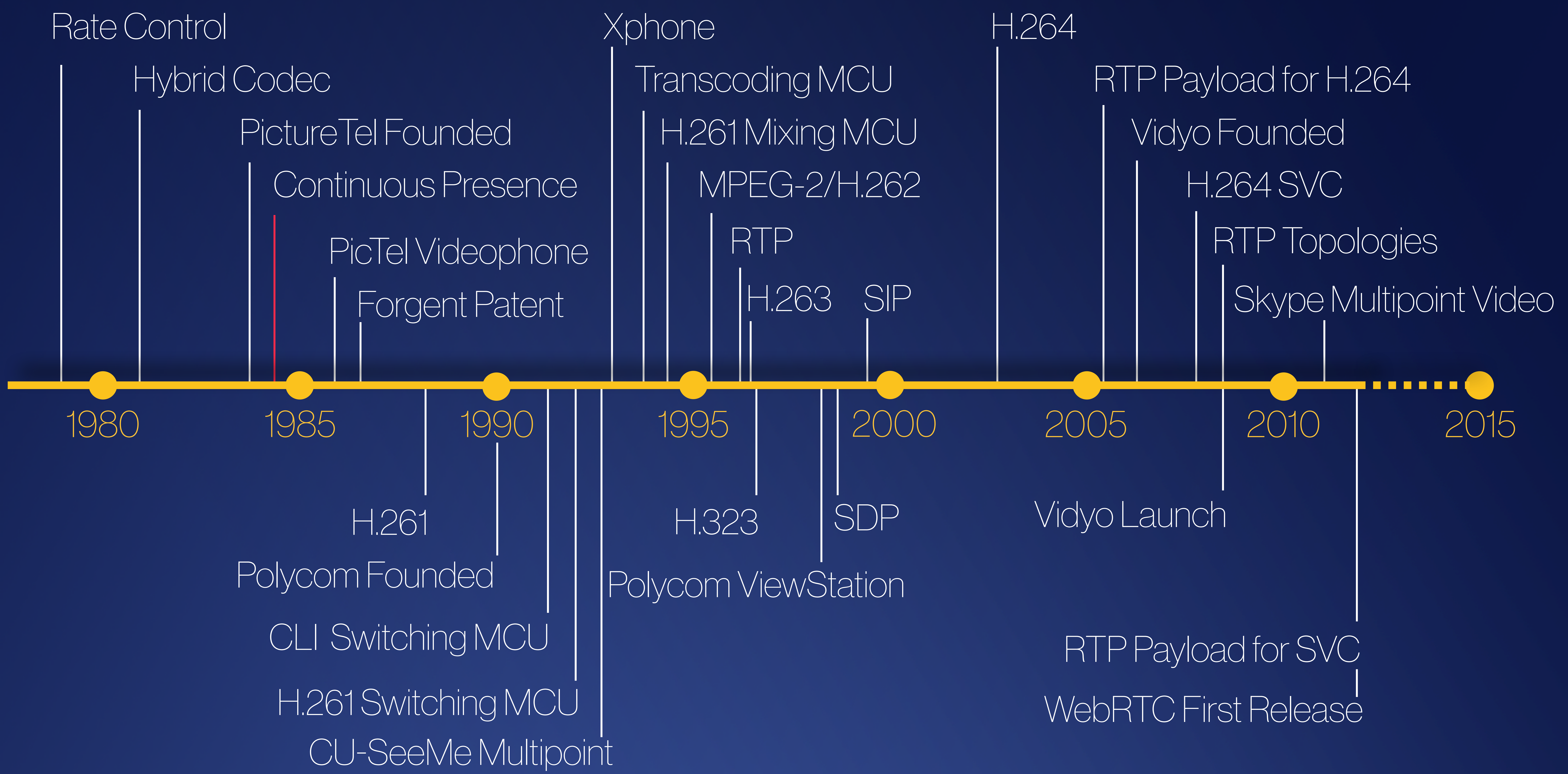
PictureTel Founded

August 1984
(as “PicTel”)
Hinman, Bernstein, and Stalin

with MIT’ers Dertouzos,
Papadopoulos, and Soley

PictureTel



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Continuous Presence

April 1985

Sabri & Prasada

Proc. of the IEEE

P2P “continuous presence” using multiplexing of multiple cameras (like telepresence) or switching

Video Conferencing Systems

SHAKER SABRI, MEMBER, IEEE, AND BIRENDRA PRASADA, SENIOR MEMBER, IEEE

Invited Paper

In this paper several video conferencing systems are discussed. These include single person-camera, voice-switched, split-screen, continuous presence, and virtual space systems. Special emphasis is placed on the more recent video conferencing systems, i.e., the continuous presence and the virtual space systems. The role of digital signal processing in the video conferencing environment is discussed. The interaction between service definition, the video conferencing system, and the digital signal processing requirements is highlighted.

I. Introduction

Teleconferencing systems can generally be classified into three main categories. The first category is audio conferencing where the system provides a shared aural space between participants using the telecommunication network. The second category is the visually augmented audio conferencing system. In this type of system, a shared visual space, in addition to the aural space, is provided. The visual space provides the means for displaying, modifying, and interacting with images. These images can be text, graphics, or photographic still images and can, in some cases, involve some form of animation. The third category is video conferencing. This type of system provides means for communicating live (moving) pictures of conference participants thus expanding the shared visual space. It also subsumes the first and second categories.

Each of the main categories; namely, audio, visual, and video conferencing, is generic. In each main category a number of different services can be supported.

This paper deals with video conferencing systems, i.e., the systems that support an enhanced visual space which includes live images of one or several participants. Audio and visually augmented audio systems are addressed in [1].

A. Historical

One of the earliest experimental video conferencing systems was installed between two locations of Bell Laboratories at Murray Hill, NJ and Holmdel, NJ [2]. The Bell Laboratories system used one full-duplex analog TV channel for transmission. Each of the conference rooms had three cameras to cover the participants. A voice-switching strategy was used to transmit the output of the camera covering the speaker to the other site. When nobody was speaking at a site the output of a fourth camera with a wide-angle lens covering all the participants was transmitted.

The Bell Laboratories system was followed by several experimental systems which included the CONFRAVISION system of British Telecom Research [3], Australian Post Offices split-screen system [4], Bell Canada's system which used voice-switched as well as split-screen systems in different locations, systems at NTT [5], and other telephone administrations. All these systems utilized analog TV transmission technology and standard TV cameras and monitors.

These were followed by video conferencing systems based on digital transmission. Bell Canada utilized digital transmission at 45 Mbits/s in their Montreal–Toronto link. In France, commercial video conferencing service at 2 Mbits/s began in 1976 [6]. The current European system is based on COST 211 codecs jointly designed and standardized by 10 European countries [7], [8]. The 2-Mbit/s codec and satellite-based networking characterize the European systems. NTT in Japan has implemented video conferencing with 6.3-Mbit/s (and also 1.5-Mbit/s) codecs and digital transmission facilities [9]. In North America, besides the public service of AT&T [10], Bell Canada, and Telecom Canada [11], a number of private networks have emerged [12], [13]. The North American systems are operating at transmission rate of 1.5 Mbits/s.

B. Approach in the Paper

The approach followed in this paper is to focus on the generic video conferencing systems and services that are the basis of all the different systems in this fast-emerging field.

The generic video conferencing systems fall into one of the following classes:

i) *Single Person-Camera System (SPC):* A single person-camera (PC) is used to capture a view of conference participants and the resulting video signal is transmitted and displayed at the other end.

ii) *Voice-Switched System (VSW):* In this system several person-cameras are used to capture pictures of the par-

Manuscript received October 3, 1984.
S. Sabri is with Bell-Northern Research, Nuns' Island, Verdun, Que., Canada H3E 1H6.

B. Prasada is with Bell-Northern Research and INRS-Telecommunications, University of Quebec, Nuns' Island, Verdun, Que., Canada H3E 1H6.

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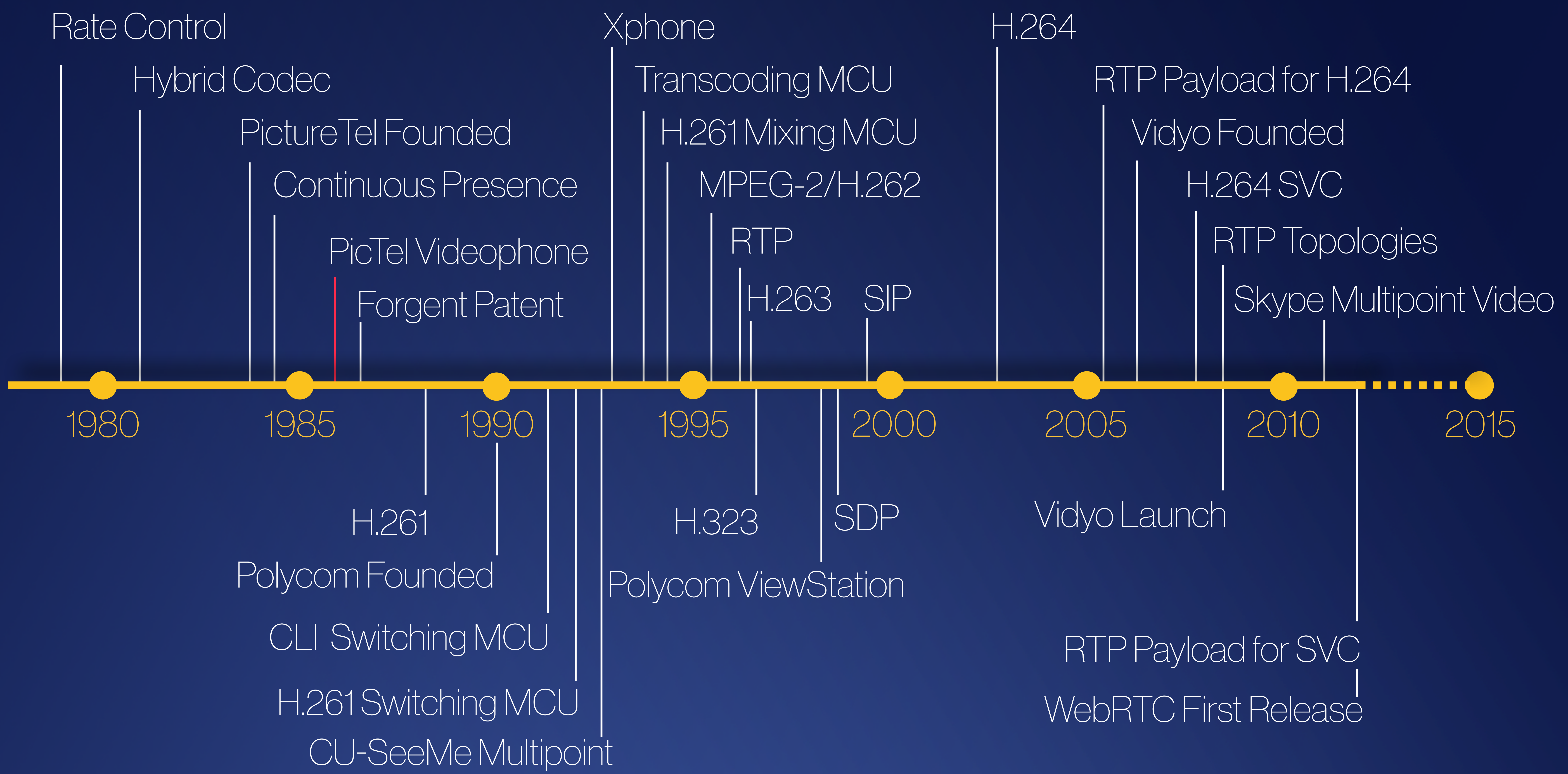

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PicTel Videophone

March 1986

2x52 Kbps
7.5 fps, 11 boards
$150K for a pair

NEW PRODUCTS

INSIDE
Software & Services/74
Microcomputers/78
Communications/79
Systems & Peripherals/83

TI adds Travelmate series terminals to Silent 700 line

Texas Instruments, Inc. of Dallas has announced the addition of the TI Silent 700 Travelmate series of portable display terminals to its Silent 700 family.

The Travelmate series includes the Travelmate, the Travelmate 1200 and the Travelmate DT. All three offer user interface modules that, according to the vendor, can be programmed to provide customized solutions for users' needs. The interface modules are plug-in application cartridges that provide 32K bytes of read-only memory and 24K bytes of random-access memory per cartridge.

TI's portables can be programmed to provide customized solutions for users.

All three standard Travelmate terminals are said to feature a retractable, 16-line by 80-column LCD screen, built-in 45 char./sec. thermal printer, full-size keyboard, data communications interfaces and specific application and text editing functions.

The Silent 700 Travelmate and Travelmate 1200 were designed for portable data communications applications. They come with internal 300 and 300/1,200 bit/sec. AT&T-compatible modems, respectively. The Travelmate DT was designed for desktop applications in which the terminal has direct connection to the host computer system via the RS-232C interface said to support up to 9.6K bit/sec. communications.

The Travelmate terminal costs $1,095; the Travelmate 1200 lists for $1,295; and the Travelmate DT costs $995, the vendor stated.

Videophone system bows from Pictel

Pictel Corp. of Peabody, Mass., recently introduced a videophone system incorporating its proprietary video image compression technology.

The system is said to include Minx videophones and a network controller manufactured by Datapoint Corp. According to the vendor, the system can support seven terminals in a star configuration network.

Pictel executives claimed that their proprietary video coder/decoder computer will transmit high-quality images and voice over low-cost 56K bit/sec. transmission lines, such as AT&T's Accunet Switched 56 Service.

A spokesman stated that Pictel's Motion Compensated Transform compression algorithm measures and compensates for changes in motion, reducing the system's need to regenerate images of moving objects.

Voice is transmitted over a separate audio line.

Software able to be upgraded

According to a spokesman, Pictel offers an advantage over earlier videoconferencing products by being software upgradeable. With the Pictel system, users will be able to plug in a software card to gain future enhancements, he explained.

The commercial product will operate at 52K bit/sec. over full-duplex digital lines and will operate at faster speeds with improved quality over dedicated and T1 lines.

The product will be available by the end of the year at a cost of $150,000 for a system including five Datapoint terminals, a local-area network connecting terminals within a building and the video coder/decoder.

The Datapoint terminals accommodate IBM Personal Computers and compatibles, but the system is not currently capable of transmitting data, the spokesman said. He added that graphics can be compiled on a micro, and pictures of the graphics can be transmitted.

DEC introduces LG family

Two line matrix printers available for text, graphics

Digital Equipment Corp. of Maynard, Mass., has introduced the LG family of printers, including the LG01 for text and the LG02 for text and graphics.

The LG01 and the LG02 are said to be 600 line/min line matrix printers. According to the vendor, the LG01 offers data processing and correspondence-mode printing as well as multiple char./in. print selections. It is said to be compatible with both existing U.S. and European character sets.

The LG02 text/graphics printer reportedly offers the capability to create bar codes, custom forms and logos; superscript and subscript modes; and prints in landscape mode. In addition, the LG02 printer provides all of the text capabilities of the LG01 printer, the vendor stated.

According to a spokesman, the number of moving, mechanical parts of both printers has been reduced, providing electrical reliability. The vendor also claimed that the printers require no scheduled maintenance calls.

Both printers are said to be compatible with DEC VAX computers from the VAX 8600 to Microvax II systems as well as with the company's Micro/PDP-11/73, Micro/PDP-11/83 and Micro/PDP-11/84 computer systems.

The LG01 costs $11,950. The LG02 costs $14,000.

True Basic's enhanced language supports Hercules' graphics card

True Basic, Inc. of Hanover, N.H., has released a new version of the True Basic Language System to support advanced graphics for the Hercules Computer Technology Co. monochrome graphics card.

True Basic's Hercules support is said to incorporate all the features of the standard True Basic language system. According to a spokesman, True Basic is the first language that allows Hercules graphics users to access 640K bytes of memory and the Intel Corp. 8087 processor.

Using True Basic with the Hercules monochrome card, users can de- ... the vendor.

True Basic's graphics syntax is said to be hardware independent, similar to the Graphical Kernel System graphics standard, so graphics developed using the Hercules card are directly portable to systems supported by True Basic's other adaptations for the IBM Color Graphics Adapter, Apple Computer, Inc. Macintosh and the Commodore Business Machines, Inc. Amiga.

The Hercules support package is available as an optional upgrade to registered True Basic users for $39.95.

HP library implements GKS

Hewlett-Packard Co. of Palo Alto, Calif., has introduced a two-dimensional graphics library that is said to be an implementation of Level 2B of the industry-standard Graphical Kernel System (GKS).

The HP-GKS library was designed for use with HP 9000 technical computers and graphics hardware under the HP-UX operating system.

It is a tool that applications developers can use in drafting, process control, mapping, simulation analysis and presentation graphics. The applications developed may achieve performance up to 50,000 2-D vector/sec. on high-performance graphics display hardware.

... segment operations that can be used to create, copy, associate and delete segments; metafile input and output defined using the GKS metafile format; attribute binding that allows the attributes to be specified individually for each primitive or bundled together; tables that can be defined for general use; and input operations that are available for six classes of input in request mode.

HP-GKS is said to support a dynamic workstation configuration and provide device-independent access to graphics workstations. In addition, applications that have already been written to the GKS standard are portable to the HP 9000 systems.

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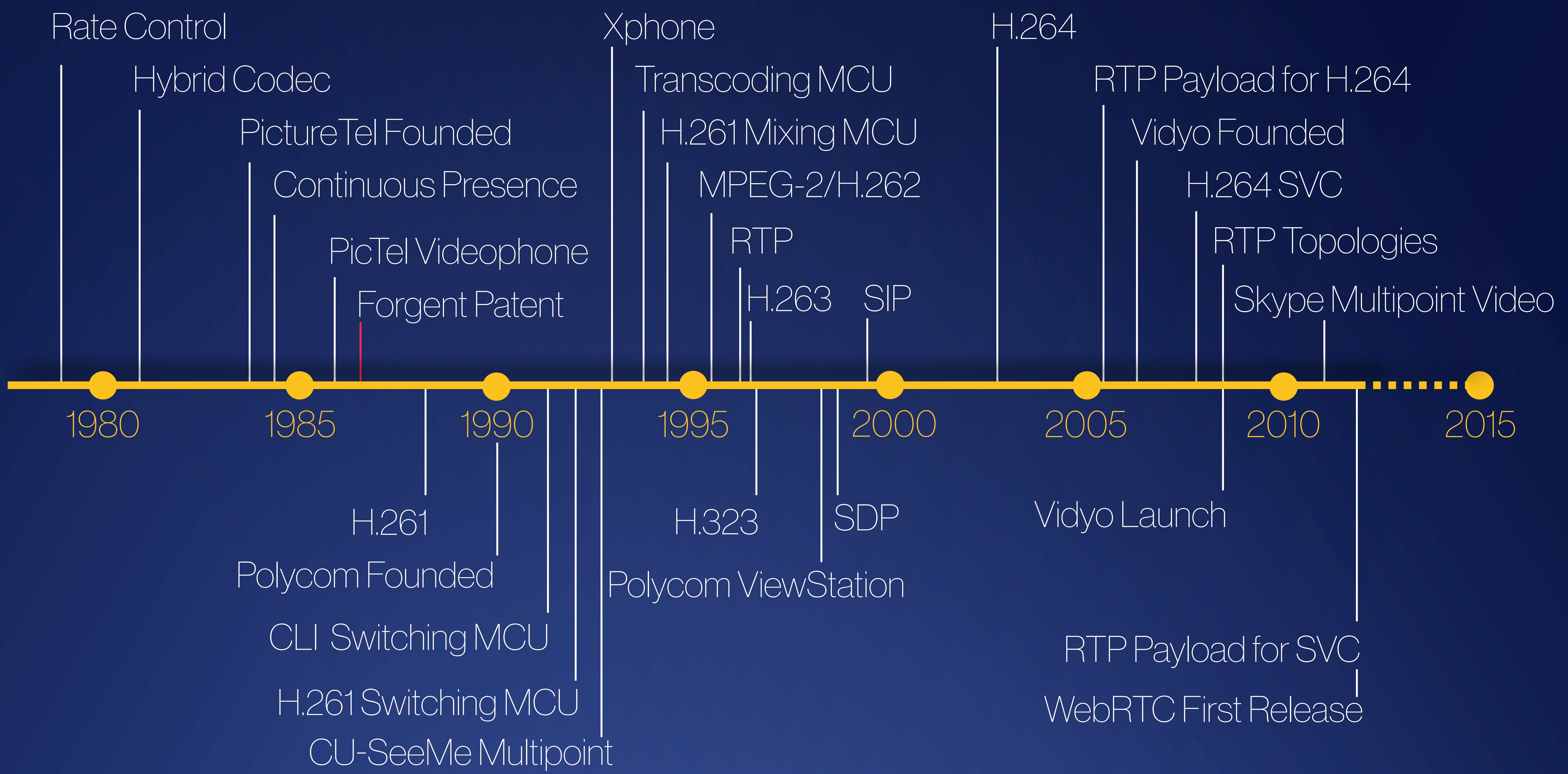
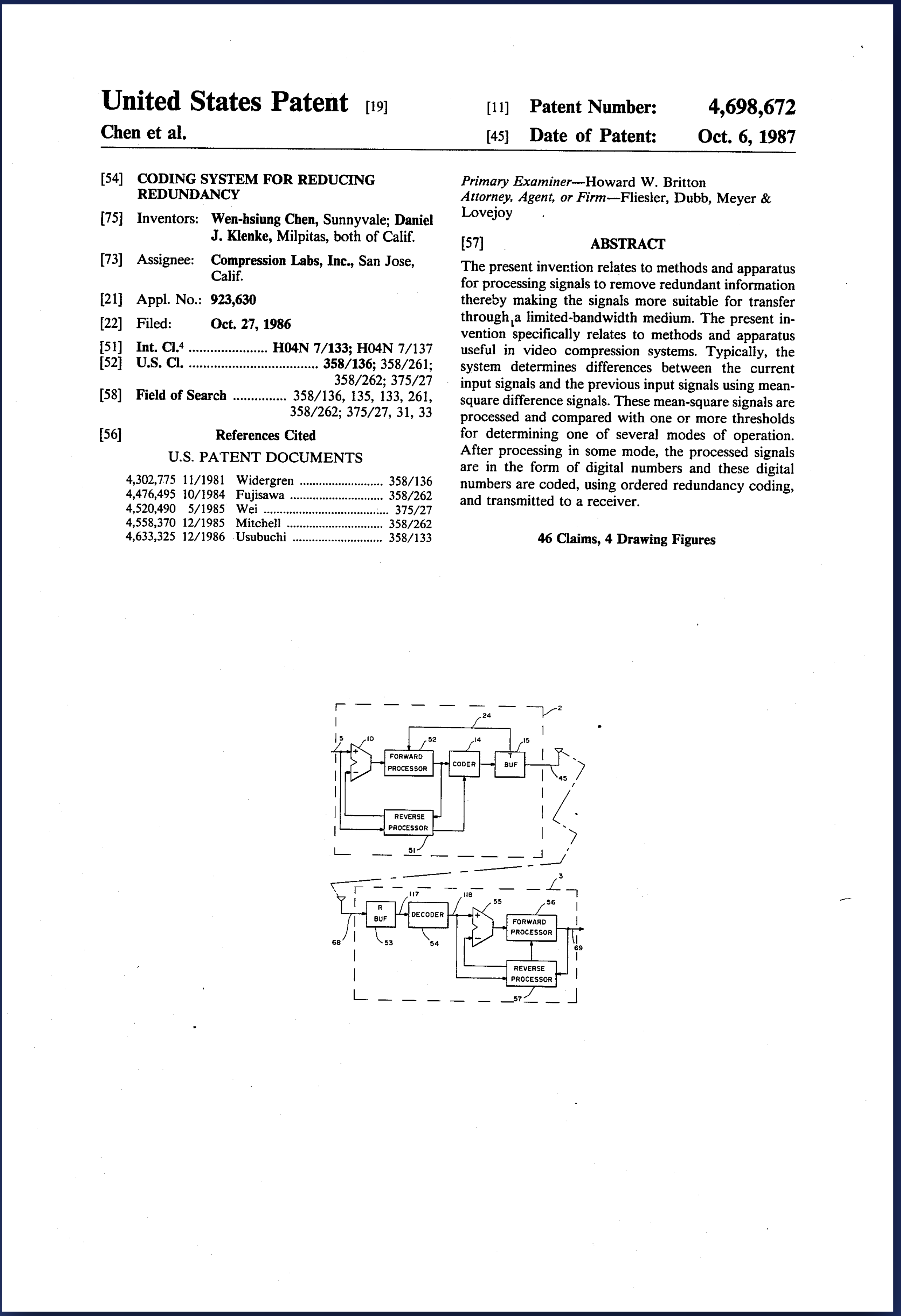

Rate Control
Hybrid Codec
PictureTel Founded
Continuous Presence
PicTel Videophone
Forgent Patent
Xphone
Transcoding MCU
H.261 Mixing MCU
MPEG-2/H.262
RTP
H.263
SIP
H.264
RTP Payload for H.264
Vidyo Founded
H.264 SVC
RTP Topologies
Skype Multipoint Video
1980
1985
1990
1995
2000
2005
2010
2015
H.261
Polycom Founded
CLI Switching MCU
H.261 Switching MCU
CU-SeeMe Multipoint
H.323
SDP
Polycom ViewStation
Vidyo Launch
RTP Payload for SVC
WebRTC First Release

Forgent Patent

October 1986
US Pat. Nr. 4,698,672
Compression Labs, Inc.

"JPEG patent"



United States Patent [19]
Chen et al.

[11] **Patent Number: 4,698,672**
[45] **Date of Patent: Oct. 6, 1987**

[54] **CODING SYSTEM FOR REDUCING REDUNDANCY**

[75] Inventors: **Wen-hsiung Chen**, Sunnyvale; **Daniel J. Klenke**, Milpitas, both of Calif.

[73] Assignee: **Compression Labs, Inc.**, San Jose, Calif.

[21] Appl. No.: **923,630**

[22] Filed: **Oct. 27, 1986**

[51] **Int. Cl.**[4] **H04N 7/133**; H04N 7/137

[52] **U.S. Cl.** **358/136**; 358/261; 358/262; 375/27

[58] **Field of Search** 358/136, 135, 133, 261, 358/262; 375/27, 31, 33

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4,520,490	5/1985	Wei	375/27
4,558,370	12/1985	Mitchell	358/262
4,633,325	12/1986	Usubuchi	358/133

Primary Examiner—Howard W. Britton
Attorney, Agent, or Firm—Fliesler, Dubb, Meyer & Lovejoy

[57] **ABSTRACT**

The present invention relates to methods and apparatus for processing signals to remove redundant information thereby making the signals more suitable for transfer through a limited-bandwidth medium. The present invention specifically relates to methods and apparatus useful in video compression systems. Typically, the system determines differences between the current input signals and the previous input signals using mean-square difference signals. These mean-square signals are processed and compared with one or more thresholds for determining one of several modes of operation. After processing in some mode, the processed signals are in the form of digital numbers and these digital numbers are coded, using ordered redundancy coding, and transmitted to a receiver.

46 Claims, 4 Drawing Figures

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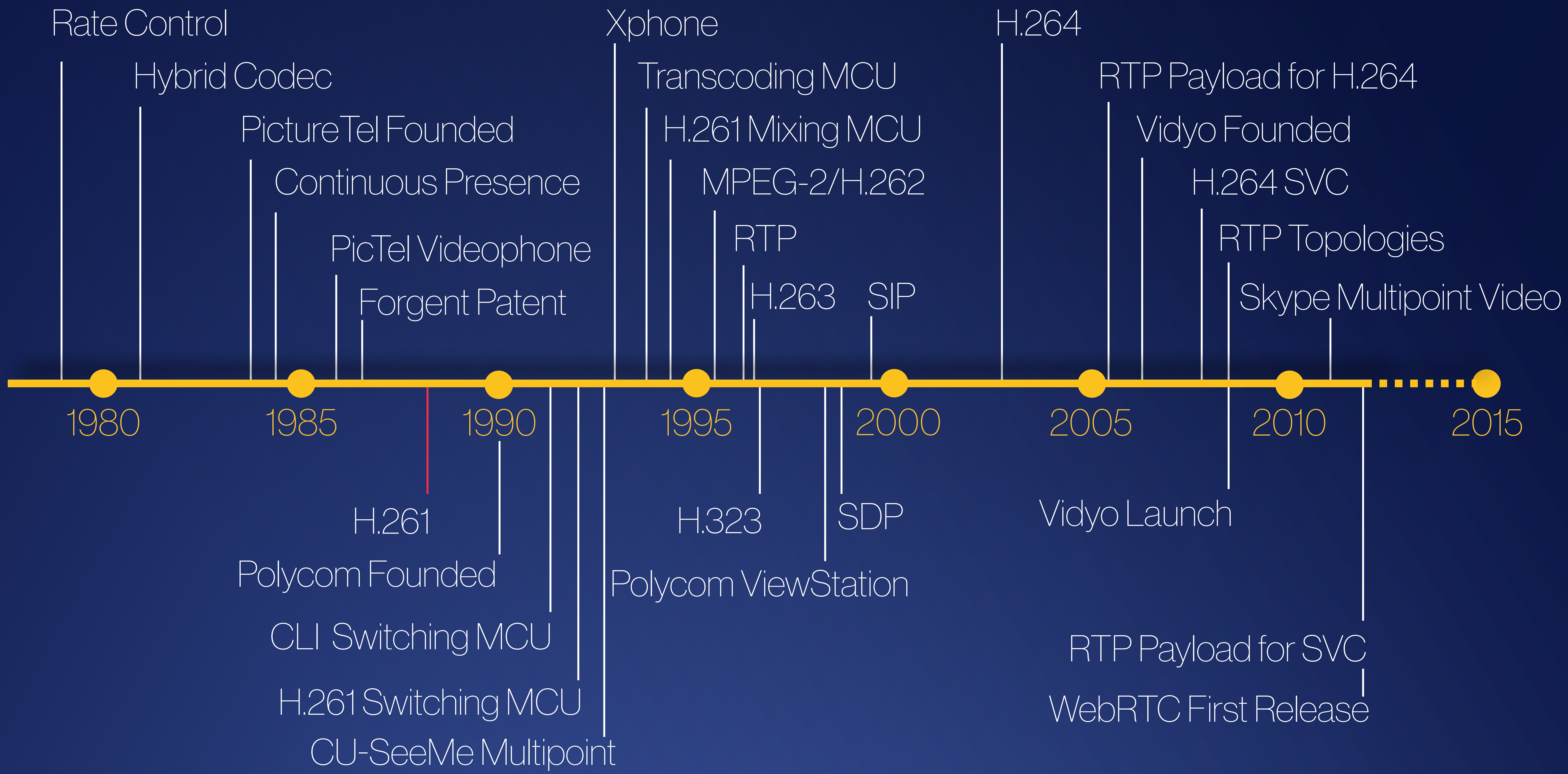
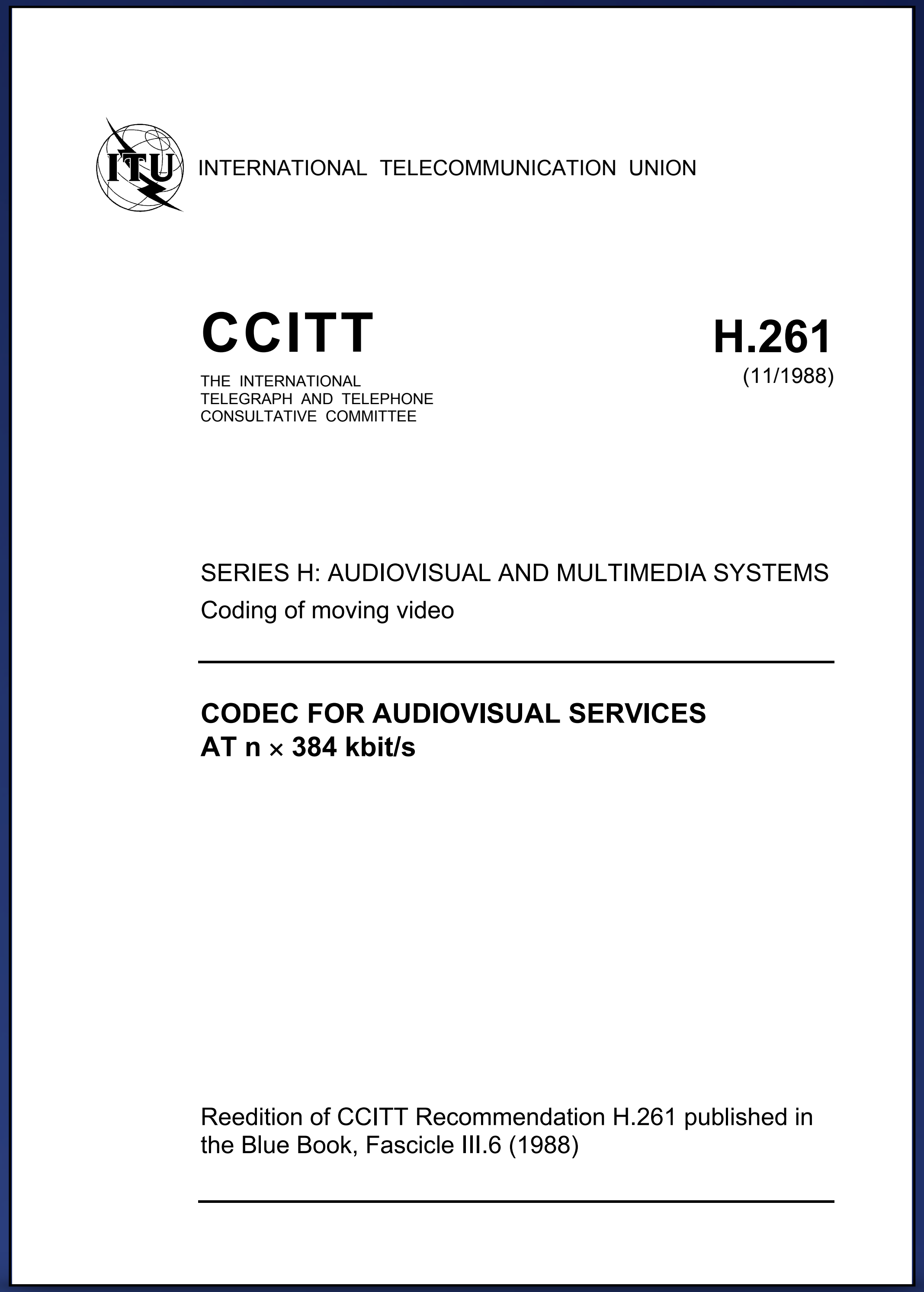


Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

H.261

November 1988

First digital video compression standard for communications


INTERNATIONAL TELECOMMUNICATION UNION

CCITT

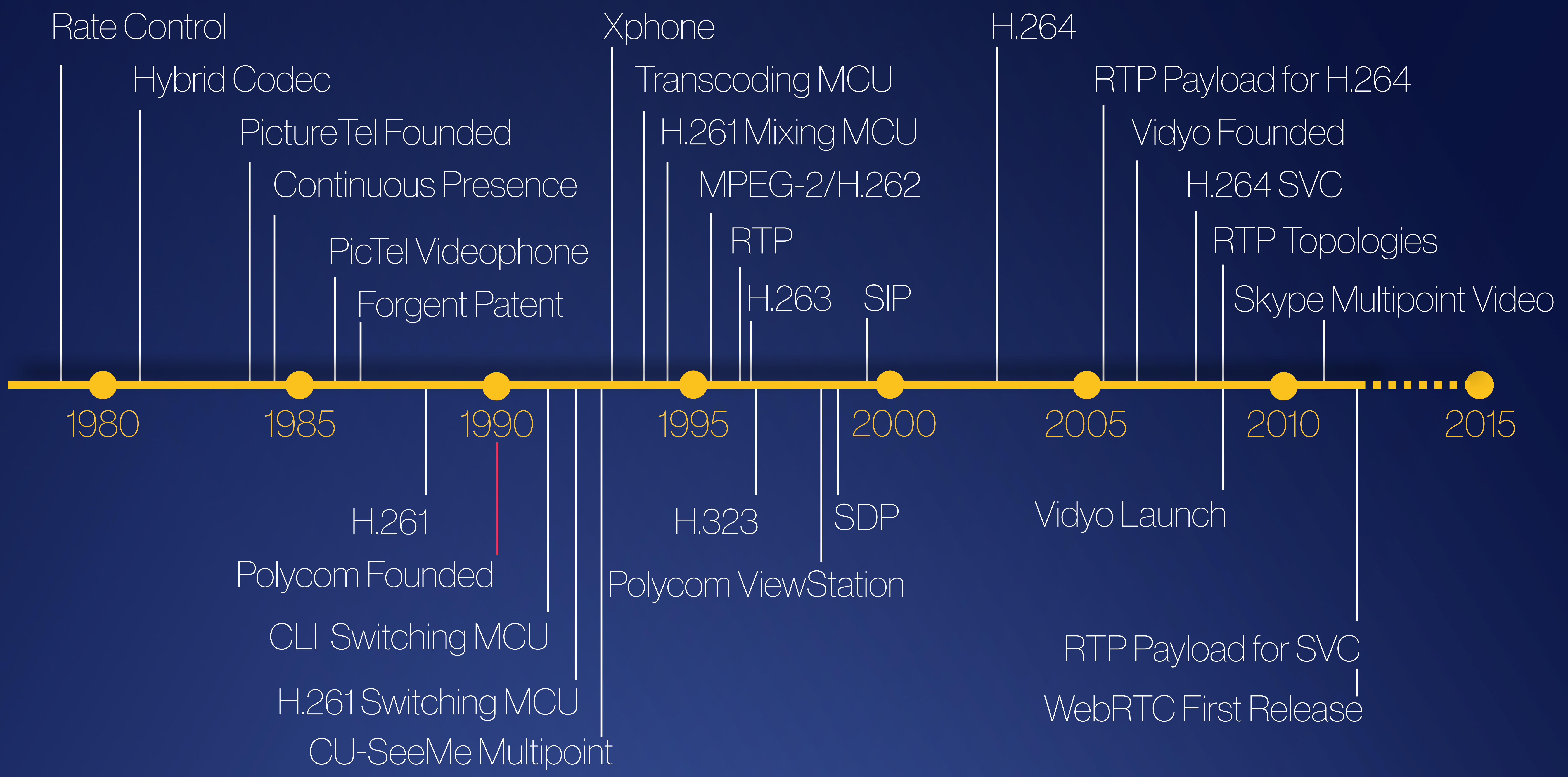
THE INTERNATIONAL TELEGRAPH AND TELEPHONE CONSULTATIVE COMMITTEE

H.261
(11/1988)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS
Coding of moving video

CODEC FOR AUDIOVISUAL SERVICES AT n × 384 kbit/s

Reedition of CCITT Recommendation H.261 published in the Blue Book, Fascicle III.6 (1988)



Vidyo

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Chief Scientist & Co-founder

Polycom Founded

January 1990
Hinman and Rodman

POLYCOM®

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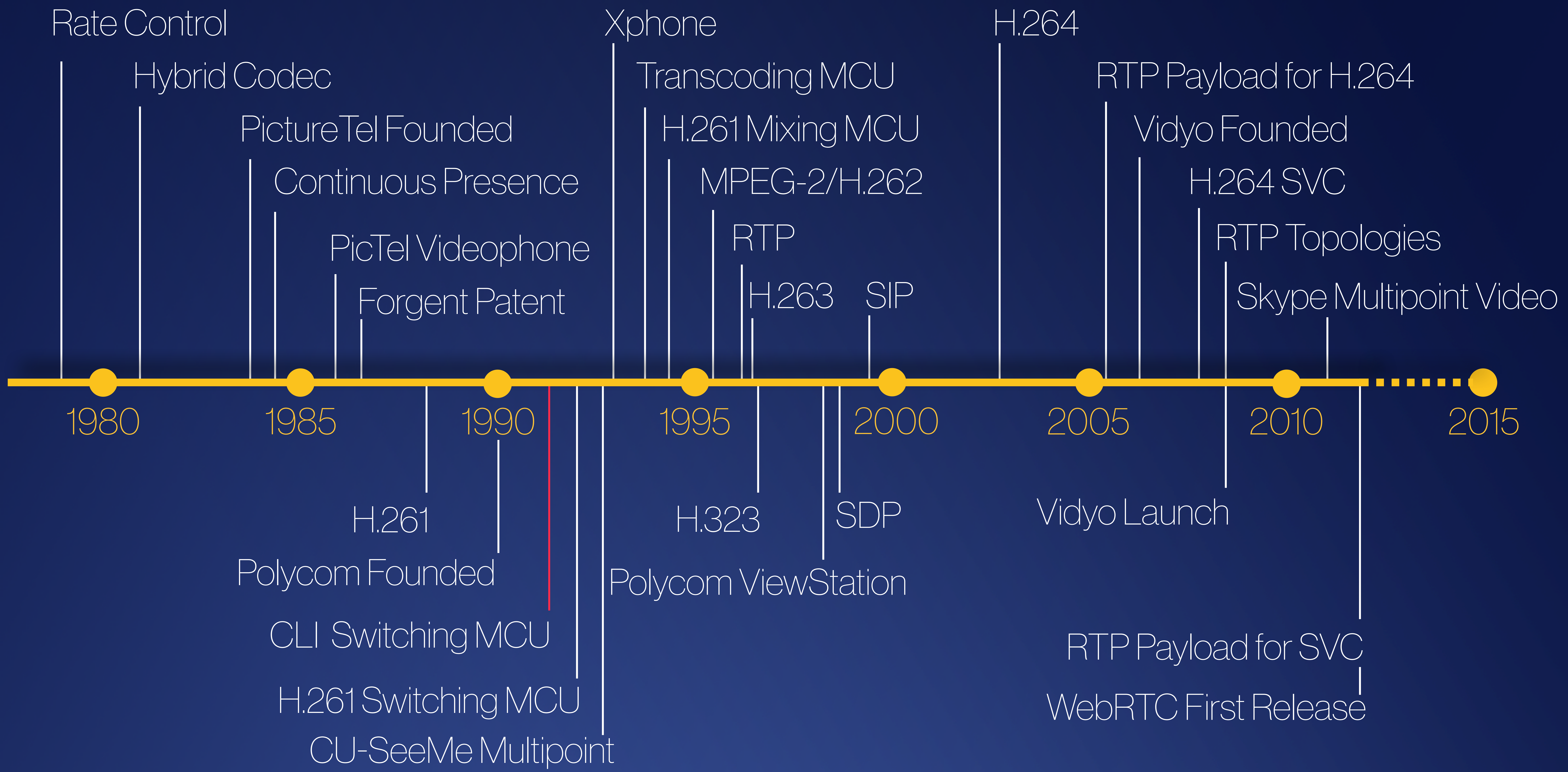
Chief Scientist & Co-founder



Vidyo

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CLI Switching MCU

February 1991

8-way
or 14-way with
cascading

MANAGEMENT STRATEGIES

MANAGING PEOPLE AND TECHNOLOGY: USER GROUPS AND ASSOCIATIONS

Worth Noting

"For almost all companies, communications networks are as much a competitive tool as having a modern factory building."

Ron Brown
Telecommunications/networking consultant
Melrose, Mass.

Association Watch

Timeplex, Inc. will hold its first Users Network Conference from Feb. 27 to March 1 in Orlando, Fla. The conference is designed to give Timeplex's North American customers the opportunity to shape the company's product direction and strategies.

Timothy Zerbiec, vice-president of advanced technology at Timeplex, will deliver the keynote address on "Technology Trends in the '90s." The conference will also feature discussions on network and bandwidth management, as well as local-area network internetworking and disaster recovery.

Sessions are planned for Timeplex's LINK+ multiplexers, TimePac packet switches, TimePath cell-relay products and Time/LAN internetworking Fiber Distributed Data Interface bridges, routers and concentrators. For information, contact Jo Ann Turli at (201) 573-6475.

Service Systems International, Ltd. recently announced the formation of the S2000 Service Management System Users Group, which held its first meeting last fall in Overland Park, Kan.

At the meeting, Service Systems unveiled Release 3.0 of the wide-area version of its S2000 for the IBM Application System/400. The software is a service management package that automates most of the daily functions of a service operation.

Service Systems can be reached at (913) 661-0190.

Survey finds some workers unclear on company ethics

Respondents split on acceptance of some gifts.

By Bob Wallace
Senior Editor

HOUSTON — An informal survey of attendees at the recent International Communications Association (ICA) Winter Seminar here revealed that although users and vendors must abide by comprehensive codes of ethics, some of those policies are not fully understood.

A total of 71 users and vendors answered all or part of the 11-question survey, which was conducted by Mark Smith, deputy general counsel for Northwest Mutual Life Insurance Co. of Milwaukee. Smith stressed that the survey was not an official ICA undertaking.

"We wanted to give attendees an idea of where [users and vendors] stood as far as the ethics process and corporate requirements are concerned," Smith said. Respondents were not required to sign the questionnaires.

Attendees were asked if their companies have a code of ethics, set of guidelines or policy statement relating to business conduct and conflicts of interest. Sixty-three respondents said their firms follow such guidelines, seven indicated they did not know, and one said no such formal program existed at his firm.

Fifty-seven attendees said their company's code or statement addresses employee acceptance of gifts, travel, entertainment, fees or compensation from firms doing business or seeking to do business with their company. Three said their guidelines address these areas, and five said they did not know.

Of those responses, 18 said acceptance of items is absolutely prohibited, while 40 said they are allowed provided they fall within certain guidelines.

Most respondents said restrictions or limitations are placed on what may be accepted in terms of type, value or quantity. Many users said they are permitted to accept items less than $10, $15 or $25 in value.

(continued on page 18)

GUIDELINES
BY WAYNE ECKERSON

Establishing trust helps projects succeed

More than ever, network managers are being required to cost-justify every project they want to undertake. Not surprisingly, many strategic projects never get off the ground because the benefits they provide cannot easily be quantified in dollars and cents.

Projects that generate soft benefits — such as increases in productivity, enhanced customer service or improved quality — usually provide significant paybacks. Companies that fail to fund these strategic projects save money in the short term but sacrifice their long-term competitiveness.

Firms that apply technology in innovative ways often have senior executives who are visionary and willing to take risks. They know that technology, if applied properly, can position their company to be effective competitors in the long run.

Too few executives, however, have that degree of confidence in information technology. Many have been burned by technology projects in the past that have promised much and delivered little.

Not surprisingly, some executives are extremely reluctant to risk their reputations and precious corporate resources on another strategic technology scheme.

According to Bud Mathaisel, executive director of Ernst & Young's Center for Information Technology and Strategy in Boston, it all boils down to the credibility of those backing the project.

"Executives will approve projects if they trust the people behind them and those people have a good track record," said Mathaisel, who was formerly chief information officer at The Walt Disney Co.

(continued on page 18)

Bendix/King videoconferencing room in Olathe, Kan.

Videoconferencing propels Bendix/King

Aerospace firm relies on video sessions to keep departments, suppliers and customers in sync.

By Maureen Molloy
Staff Writer

OLATHE, Kan. — Bendix/King, a manufacturer of aircraft radios and flight instruments, is using videoconferencing to gain a competitive edge in its aerospace programs by streamlining manufacturing operations and coordinating the flow of goods from suppliers.

The use of videoconferencing allows design engineers at sites here, in Florida and the Far East to confer on an as-needed basis, giving the company a strategic edge over competitors by significantly trimming product development time.

"Videoconferencing is a breakthrough technology that's changed the way our company does business," said Larry Ehlers, General Aviation Avionics Division manager of plant engineering. "Quicker and better communications with our teaming partners, customers and vendors was the key motivation for installing videoconferencing; reduced travel was only a minor consideration."

In one instance, the network enabled Bendix/King to speed up development of a Traffic Alert and Collision-Avoidance System (TCAS) for which several airlines had asked different manufacturers to submit prototypes. The product's design was completed a year before major rivals' products, thereby allowing Bendix to capture nearly 60% of the $1 billion TCAS market.

Ehlers said Bendix/King uses videoconferencing throughout the development phase and has begun using it to coordinate design and manufacturing operations as well.

Videoconferences are held three times a month with management teams in Singapore to coordinate manufacturing operations and to design the next generation of airborne communications and navigation equipment. Monthly videoconferences are also held between Bendix/King and its two largest customers, The Boeing Co. and McDonnell Douglas Corp.

Larry Ehlers

Videoconferencing is now used for software demonstrations for the company's data processing and maintenance departments. Finance, project and inventory reviews are also completed via videoconferences. Ehlers said this frees up time and money. While the division president previously traveled to Kansas and Florida for two separate reviews, both reviews can now be completed from a single location.

Bendix/King, a division of Allied-Signal Aerospace Co., was the first organization to install

(continued on page 18)

Alex Eleftheriadis, Ph. D.
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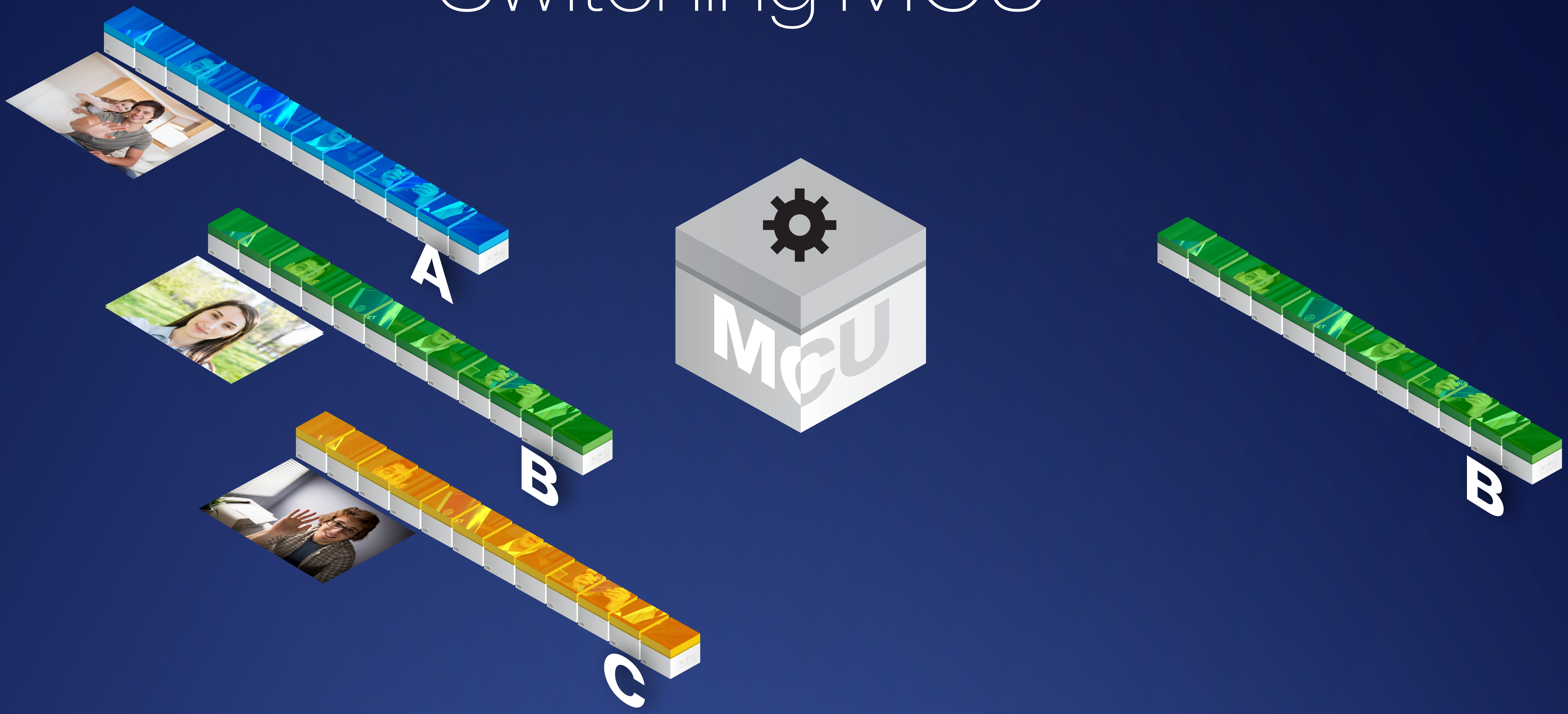
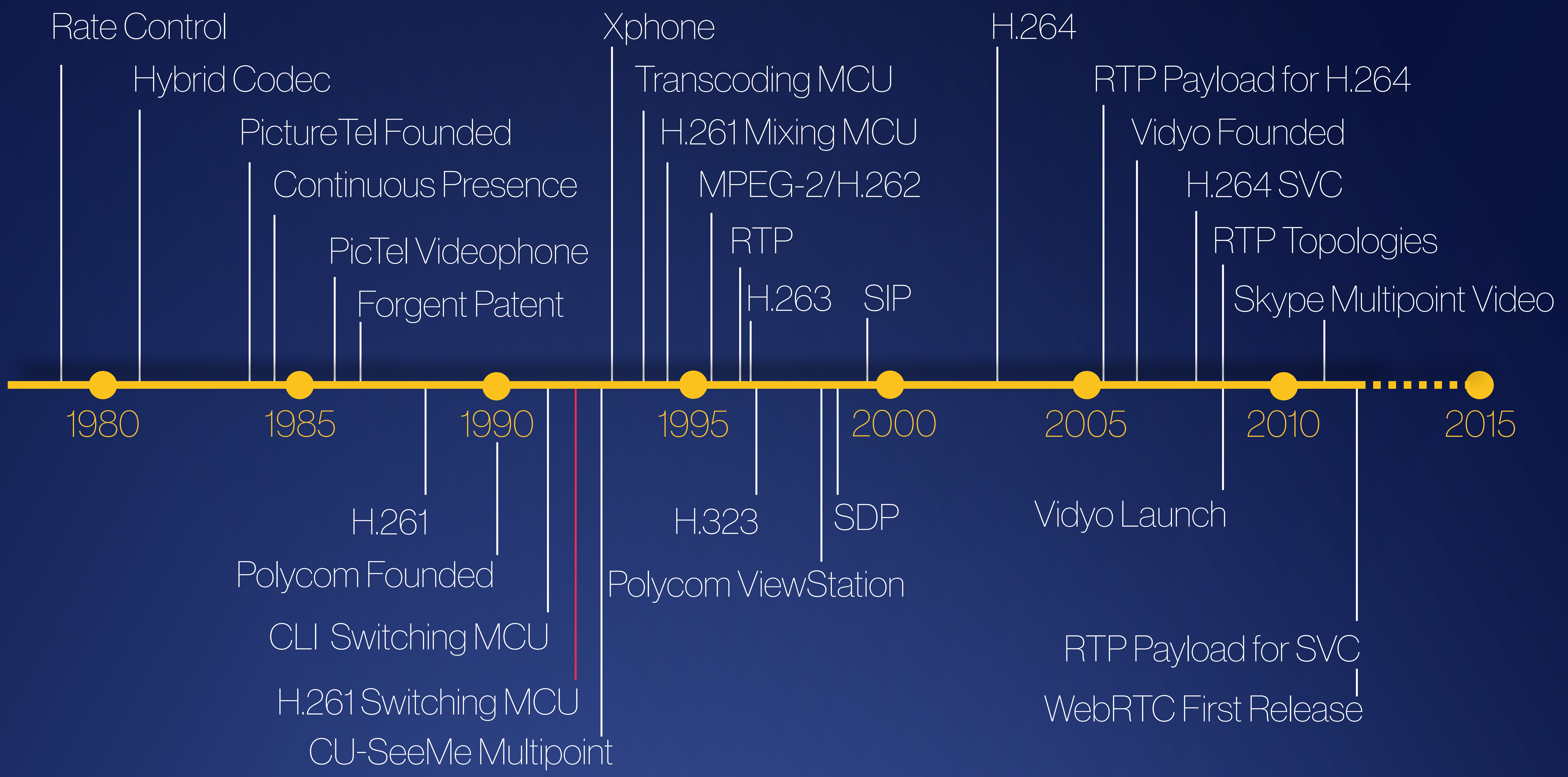
Switching MCU





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Chief Scientist & Co-founder

H.261 Switching MCU

May 1992
Clark
IEEE Comm. Magazine

Switching using H.261
(MIAS ESPRIT Project)

Multipoint Multimedia Conferencing

The expansion of ISDN and the introduction of standardized audio and video coding algorithms are accelerating the development of multipoint multimedia communications.

William J. Clark

In general terms, telecommunications systems have traditionally been designed to operate on a point-to-point basis, either between two end users or between a user and a centralized facility such as a database. Where communication is required between a number of locations (for instance, facsimile transmission between a headquarters office and a number of branch offices), this has been achieved by either a store-and-forward facility or sequential transmission from one site to each of the others. While such techniques may be quite adequate for non-real-time applications, in cases where communication between people is in progress by voice or picture, simultaneous reception of information by a number of sites is necessary. Multipoint services provide for this real-time transmission between three or more locations. The generic term "teleconferencing" is also often used, although this also covers the case where a number of people are involved on a purely point-to-point link. One of the features of teleconferencing services is that users often have a need to employ many different types of media in order to more accurately simulate a face-to-face meeting. Besides hearing and seeing other participants, they must be able to transmit documents, files, and still images, and also replicate some of the human interactions of meetings such as chairmanship.

With the increase in businesses operating on a multi-site basis, the need for standardized multipoint multimedia services is apparent. This article describes the development of multipoint multimedia services for conferencing with emphasis on the use of Integrated Services Digital Network (ISDN) as part of the work carried out in the European collaborative projects Multipoint Interactive Audiovisual Communication (MIAC) and Multipoint Interactive Audiovisual System (MIAS).

History

From the earliest days of the telephone, service operators have taken the opportunity to provide additional services. The first use of what might be called multipoint telephony took place during the last century when the sound of a live performance from the opera stage was sent to a number of subscribers. In this system, the sound was transmitted in one direction only, and due to the lack of amplification only a limited number of receivers could be used. The first multipoint audioconferences took place in the 1930s, linking housebound and hospitalized students in a school district of Iowa [1].

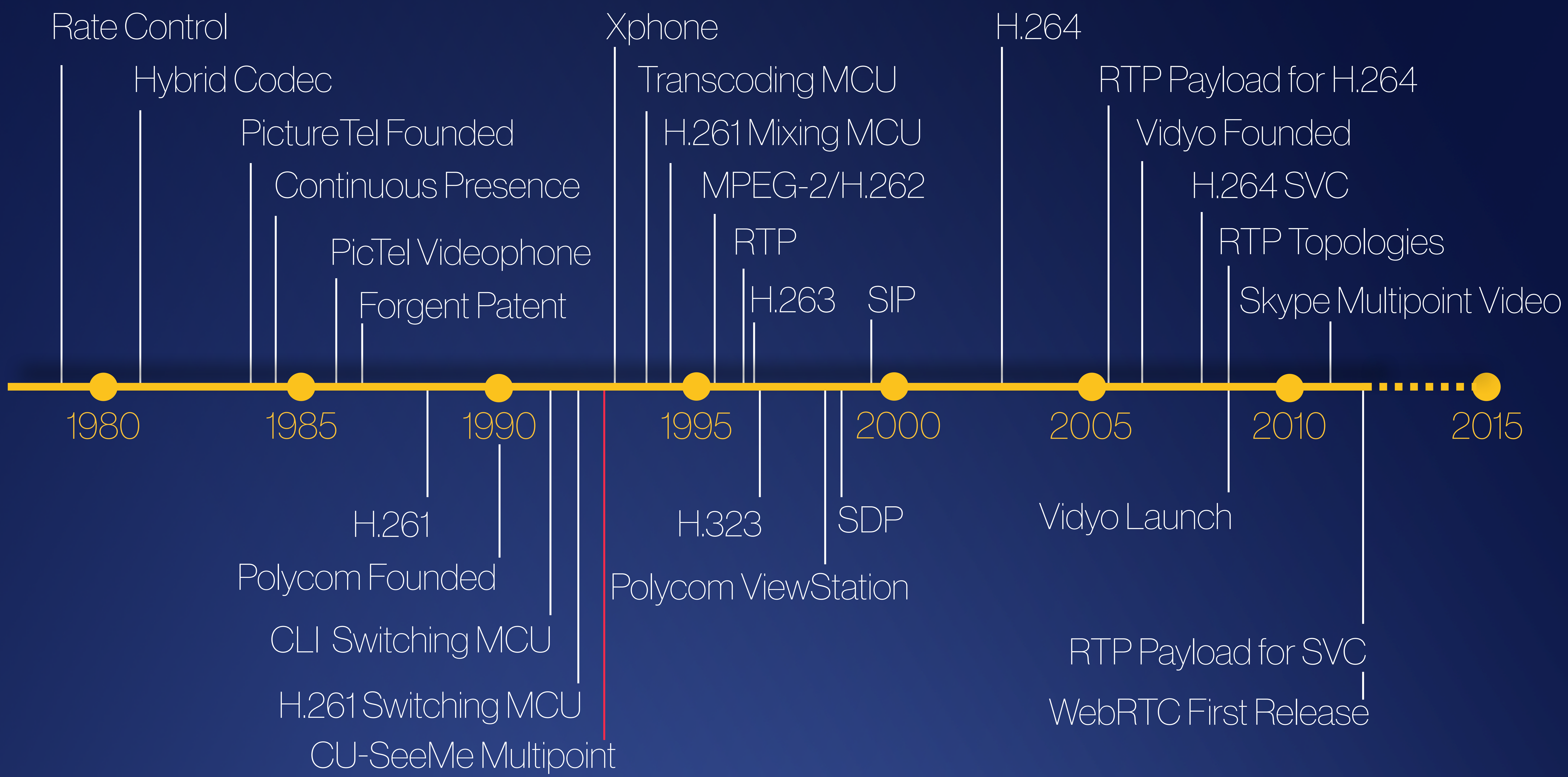
In such simple audioconferences, only audio is transmitted. Once the idea of linking a number of sites for audio had been established, it was natural to attempt to add some form of pictorial or graphical information. Basically, three types of additional information can be considered: still images, such as facsimile and slow-scan TV (now known as still-picture television); computer data, such as files; and moving video images, generally derived from a camera. The combination of these information sources with audio are known as audiographic teleconference and videoconference, respectively. For historical reasons these have tended to be considered as separate services, although, as described later, one can envisage systems providing a single service across a whole range of facilities.

Audiographic teleconferencing started in the late 1960s, using the telephone network to provide audio together with some supplementary facility. An early example of this was the use of audio and facsimile by the National Aeronautics and Space Agency (NASA) to coordinate the Apollo program. In the late 1970s, various experiments were conducted in the use of still-picture TV in conjunction with audio. In 1980, British Telecom Laboratories (BTL) implemented trials of applications for still-picture TV, including telemedicine and teleconference. Resulting from this experience, a successful system was developed, known as IMTRAN [2], which provided for the transmission of body scanner images from a hospital to a remote consultant.

In 1981, BTL with the Open University conducted a human-factor trial of the CYCLOPS system [3]. This equipment allowed users to draw interactively on the face of a television screen by means of a light pen and exchange simple images, drawings, or alphanumeric characters, together with an

WILLIAM J. CLARK heads British Telecom's Multipoint Teleconference Group.

0163-6804/92/$03.00 1992© IEEE824-7504



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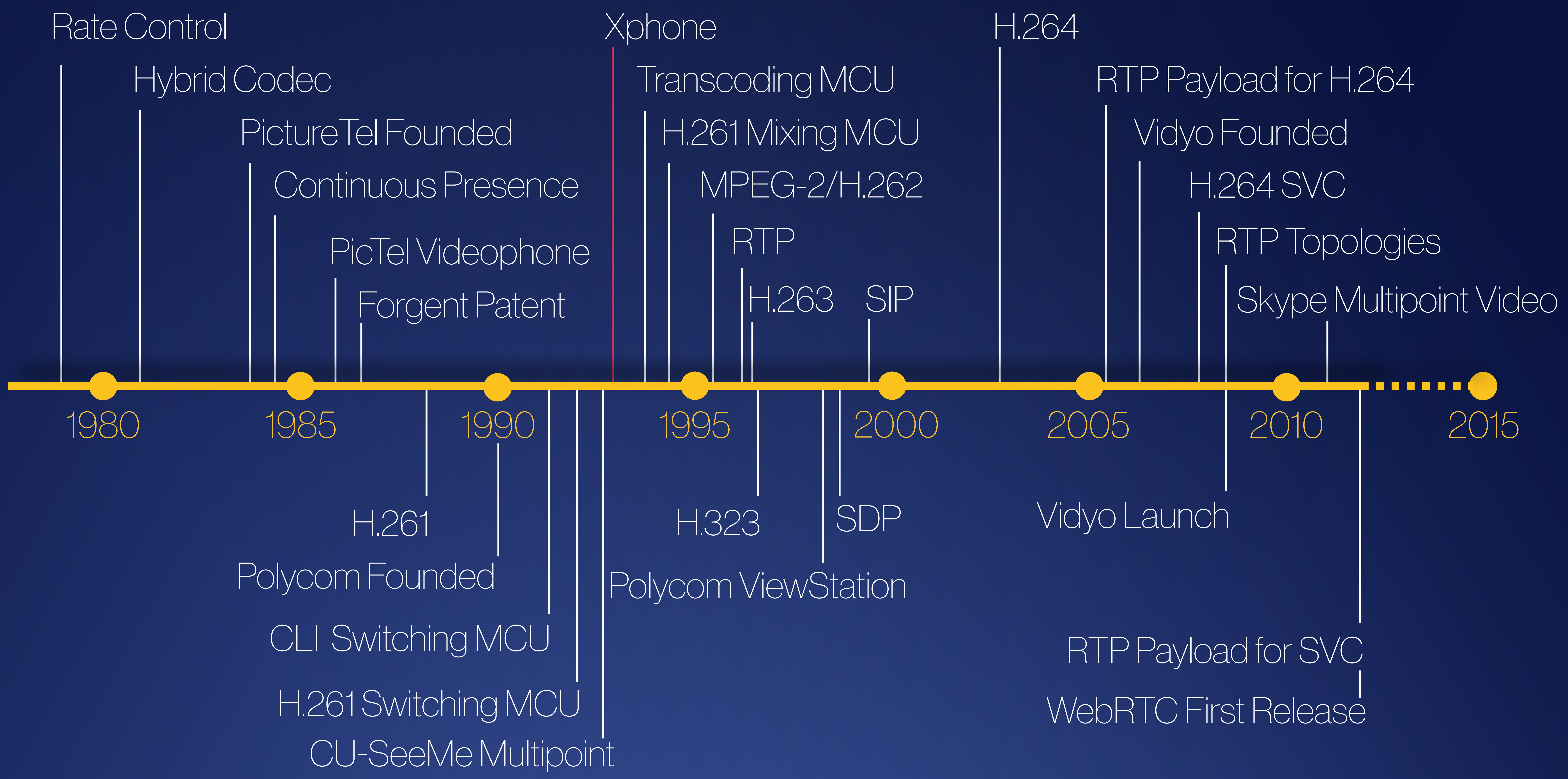
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

CU-SeeMe Multipoint

April 1993
CU-SeeMe v0.19 for Mac

Multipoint using "reflector".
First multi-stream endpoint.



Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Xphone

August 1993

Eleftheriadis, Pejhan, and Anastassiou

1st ACM Multimedia Conf.

P2P audio+video using motion-JPEG hardware on Sun workstations

Algorithms and Performance Evaluation of the Xphone Multimedia Communication System

Alexandros Eleftheriadis, Sassan Pejhan and Dimitris Anastassiou

Electrical Engineering Department, Columbia University

Abstract—**We describe and evaluate the performance of the algorithms used in Columbia University's "Xphone" multimedia communication system. The system assumes a "best-effort" operating system and network, and provides synchronized video/audio acquisition/playback (locally or across a network) with minimized and bounded end-to-end delay. Synchronization is achieved using an algorithm based on time-stamps and device state information. The effects of jitter (delay variation) are mitigated using silence detection; the end-to-end delay is kept bounded using a restart mechanism. Finally, for live video sources, we describe a source bit-rate adaptation algorithm that maximizes the video image quality to the available network bandwidth and video display window size.**

Keywords—**Multimedia communication systems, media synchronization, source rate control, application development systems.**

I. Introduction

One of the enabling technologies for multimedia systems is video compression algorithms. Recent advances in compression technology for images and video (JPEG, MPEG-1, MPEG-2) have resulted in bandwidth reductions of two orders of magnitude, down to 1–2 Mbit/sec. In addition, the work of international standardization organizations and the increased interest in video applications for computers and consumer electronics products have resulted in VLSI implementations of these algorithms which can be used for the development of real systems [1; 2; 3; 6; 13; 14; 19].

Video coding, however, is just one of the components of a multimedia system. The support of continuous, high-volume and real-time data (like video or audio) in both computers and networks represents a tremendous shift in design methodology, resulting in a re-evaluation of basic principles. Time dependency of information as a concept existed only in dedicated systems (e.g. the telephone network, or embedded systems); with multimedia, it becomes an issue for practically any application. The focal point of multimedia research is to provide bit-pipe characteristics (guaranteed bandwidth, low and constant delay, accurate synchronization) to packet-based systems, using algorithms and architectures that can be widely deployed.

The availability of some kind of real-time support from the underlying operating system and network is important for high-quality, wide-area multimedia communications, and is currently a very active area of research (see e.g. [9; 10; 20] and references therein). It is nevertheless possible to provide multimedia communication even in environments where delay uncertainty prevails (best-effort systems), albeit with some quality degradation. In addition, algorithms employed in non-real-time and real-time systems can be the same; although the latter will definitely perform better, the techniques used to achieve this performance can be similar (especially if the real-time support is not "hard"). Throughout this paper we assume the use of a best-effort operating system and network; in other words, no time-related guarantees are provided.

A number of systems and techniques have appeared in the literature, addressing various aspects of multimedia systems. Early efforts provided audio communication only [4]. Some systems use analog video and audio communication [5], with the corresponding self-evident limitations in terms of media integration in user applications. A significant volume of work has been reported at the system architecture level [8; 16; 18], describing the interface between applications and multimedia services and the latter's structure. In the area of media synchronization, a number of techniques have been proposed. These include incorporation of time constraints and scheduling of multimedia documents [7; 12; 15], media synchronization for database access applications (where a high end-to-end delay is acceptable) [8; 17], and synchronization for interactive multimedia communications [11]. In the first and second areas, the proposed techniques are basically used to derive time-stamps (or their equivalent) with no further analysis of how these time-stamps will be enforced; in addition, strong assumptions are usually made in terms of the performance of the underlying network and host equipment [17]. In the third area, which is more directly related to our work, the techniques described in [11] are only applicable in token ring networks (where the network access time is known prior to transmission) and require very tight coupling of software/hardware layers (the authors use their own operating system).

In this paper we describe the architecture and associated algorithms of the Xphone multimedia communications system, which has been developed to support the use of multimedia

Work supported in part by the New York State Center for Advanced Technology in Computers and Information Systems, and in part by the Center for Telecommunications Research at Columbia University. The authors are with the Electrical Engineering Department, Columbia University, New York, NY 10027 ({eleft,sassan,anastas}@ctr.columbia.edu).

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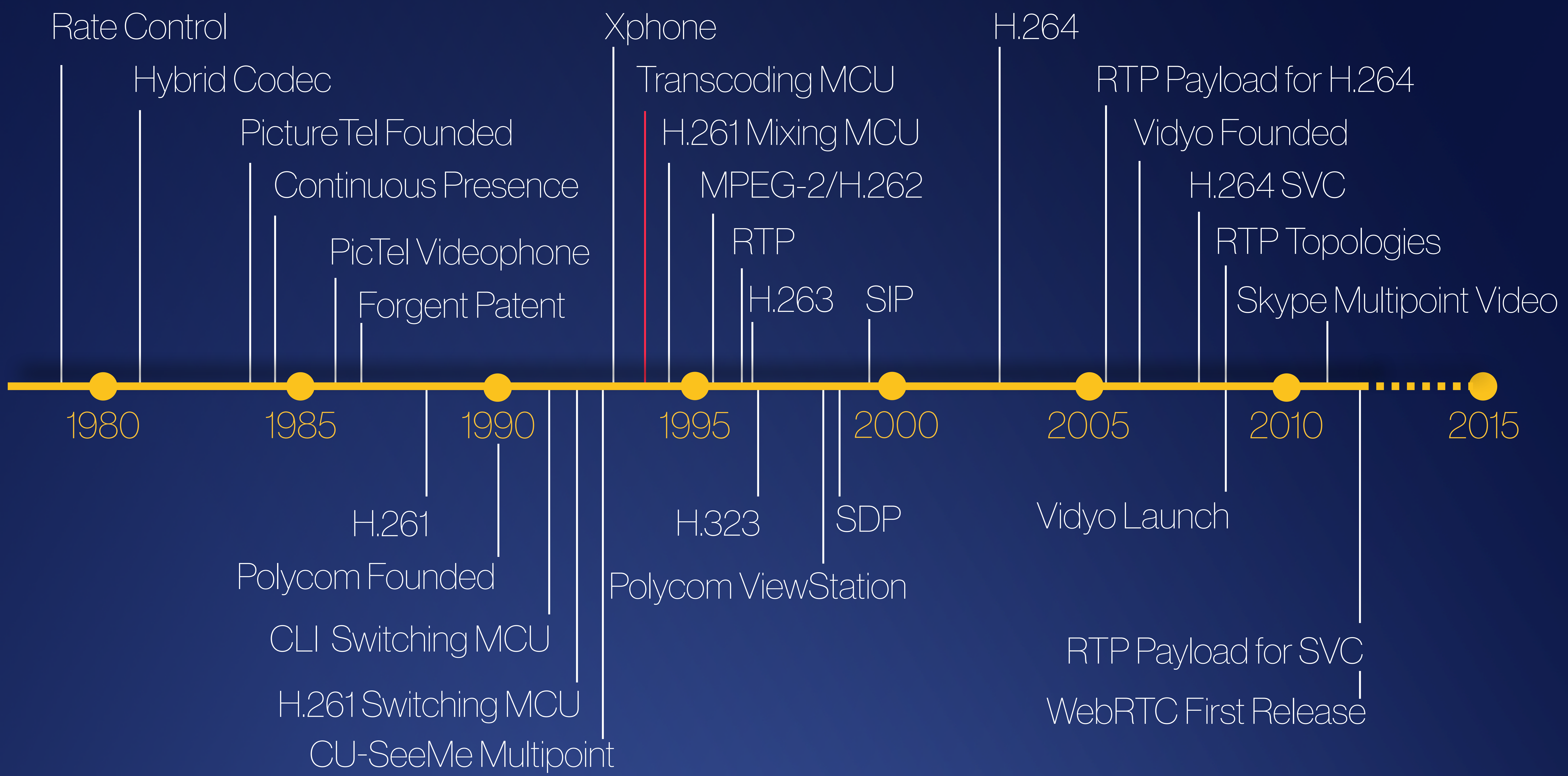

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Transcoding MCU

January 1994

Willebeek-LeMair, Kandlur, and Shae

19th Conf. on Local Computer Networks

First reference to a transcoding gateway

On Multipoint Control Units for Videoconferencing

M. H. Willebeek-LeMair, D. D. Kandlur, and Z.-Y. Shae

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Abstract

This paper examines the issues involved in the design of conference servers that support multiparty, multimedia conferences. These servers, called Multipoint Control Units (MCUs) in the telephony world, coordinate the distribution of audio, video, and data streams amongst the multiple participants in a videoconference. The MCU is responsible for the processing of video and audio so that a conference participant can hear and see one or more of the other participants in the conference. It is also responsible for handling and forwarding the data streams from the participants. This paper presents different approaches to the design of an MCU to implement these functions. It also describes the design of a related device – a transcoding gateway that enables conferencing between participants using different video/audio equipment.

1 Introduction

In recent years, with the emergence of improved communication technologies with wider coverage and accessibility, videoconferencing has become one of the major new growth applications. Videoconferencing standards are being developed and more and more videoconferencing products are appearing in the market.

Videoconferencing solutions are currently evolving from several directions. On the one side, there are the circuit-switched (e.g., Narrowband ISDN or the Switched-56Kbps phone lines) types of solutions, which being motivated by the telephony industry, can be likened to it. On the other side are the packet-based network (e.g., Ethernet and Token Ring legacy LANs) solutions, which are designed to carry real-time traffic over existing computer communications networks. The advent of ATM [1] (Asynchronous Transfer Mode) might eventually allow these two approaches to converge.

0

Due to the stringent bandwidth, delay, and jitter requirements of real-time audio and video data, the solutions for the circuit switched and packet-based networks differ considerably. These differences include the encoder/decoder (CODEC) technology for video compression and decompression, the methods used to guarantee network performance, and in the provisions within the end-stations to handle the real-time traffic.

Videoconferences may be point-to-point or multipoint.

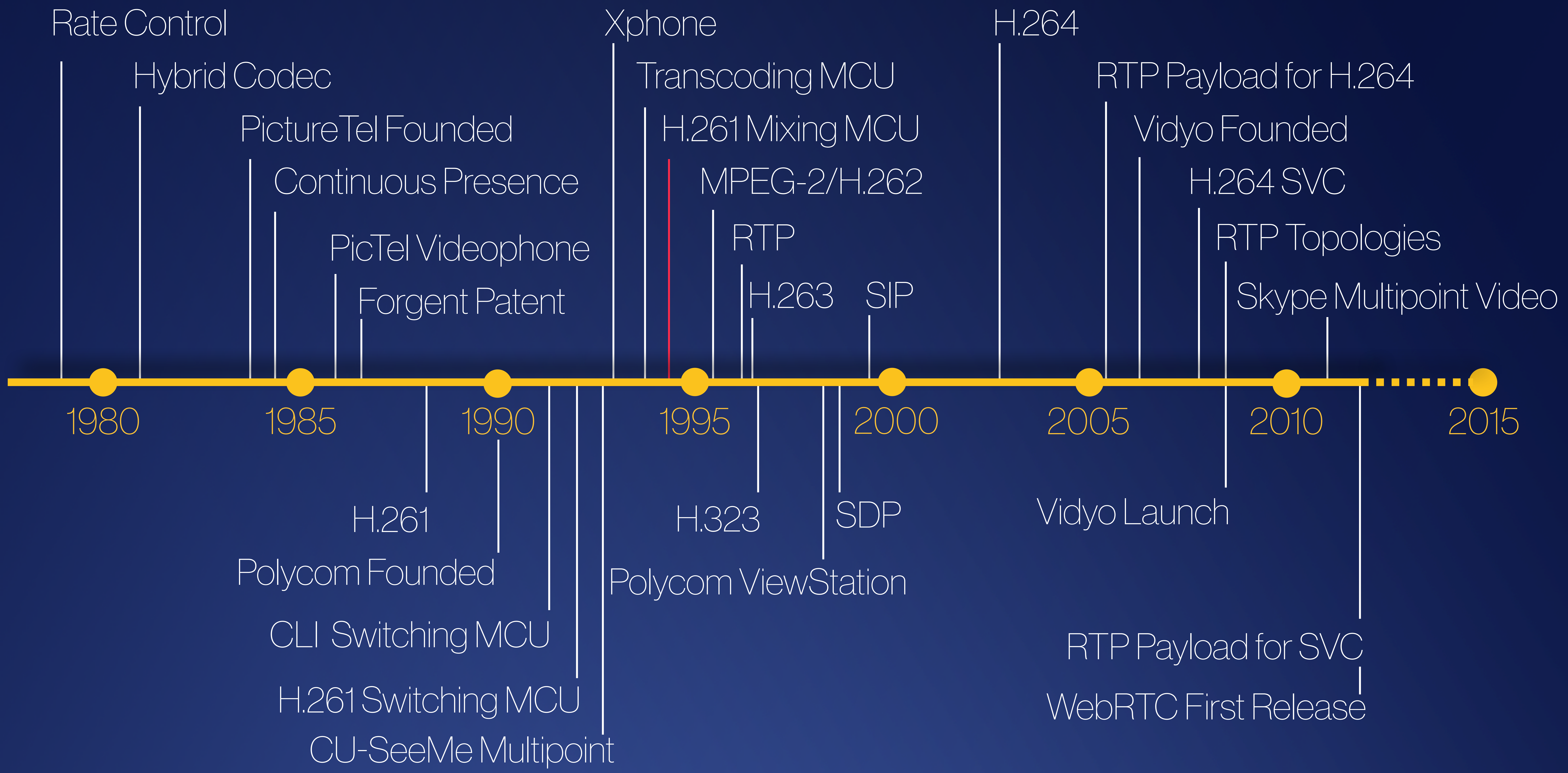
Point-to-Point. In a point-to-point videoconferencing a user is able to connect to only one other participant and communicate via video, audio, and shared data applications.

Multi-Point. A multi-point conference involves more than two participants and multimedia data is multicast from each participant to all others.

Each of the above scenarios involves the integrated communication of video, audio, graphics, and text data.

Approaches used to support multipoint conferences may be categorized as either distributed or centralized. In a distributed approach each end-station receives the video and audio streams from all, or some, of the participating end-station sources in the conference. Each end-station then composes these multiple incoming streams as desired. This approach is advantageous since it allows more flexibility and control at each end-station and minimizes the distance that streams need to travel between source and destination. It requires additional processing capability

Transcoding MCU



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H.261 Mixing MCU

August 1994
Lei, Chen, and Sun
IEEE Trans. on CSVT

H.261 "mixing" bridge
(Bellcore)

Video Bridging Based on H.261 Standard

Shaw-Min Lei, *Member, IEEE*, Ting-Chung Chen, *Senior Member, IEEE*, and Ming-Ting Sun, *Senior Member, IEEE*

Abstract— **Multi-point ISDN videoconferencing with video bridging in network-based servers represents a viable new network service. This paper presents a detailed technical analysis of a continuous presence video bridge using the H.261 video coding standard. We first compare the pros and cons of coded domain versus pel domain video bridges. The architecture and the required operations of a coded domain bridge using H.261 are then investigated. We derive the bounds of the bridge delay and the required buffer size for the implementation of the bridge. The delay and the buffer occupancy of the video bridge depend on the order, complexity, and the bit-distribution of the input video sources. To investigate a typical case, we simulate the delay and the buffer occupancy of a video bridge. We also provide a heuristic method to estimate the delay in a typical case. Several techniques are discussed to minimize the bridge delay and the buffer size. Finally, we simulate intra slice coding and show that the delay and the buffer size can be reduced significantly using this technique.**

I. Introduction

MULTI-point videoconferencing is a natural evolution of two-point videoconferencing. To establish multi-point connections, coded video streams from the participants are sent to a Multi-point Control Unit (MCU). In a "switched presence" MCU, either a signal selected by the conference chairman or a signal selected based on audio channel activity is broadcast to all participants [1]. In a "continuous presence" MCU, multiple video streams from the participants are combined so that each participant can see the selected multiple participants all the time. The "switched presence" MCU has been standardized recently [2], [3] while the "continuous presence" MCU is still under active research. In this paper, we focus on video combining in the "continuous presence" MCU to support network-based multi-point multimedia videoconferencing.

In the MCU, multiple coded video sources can be decoded, combined in the pixel domain, and then encoded for distribution. In some special cases, however, the video sources can be combined directly in the coded domain without transcoding. Video combining in the coded domain offers shorter end-to-end delay, better picture quality, and lower MCU cost. It happens that the current videophone/videoconferencing standard H.261 [4] is able to provide such video combining without any transcoding. The MCU can combine four QCIF (Quarter Common Intermediate Format) videos into one CIF video in the coded domain for continuous presence multipoint videoconferencing. Each user terminal is operated in an asymmetric mode which transmits QCIF pictures but receives CIF pictures at four times of transmission bit rate. A QCIF video combiner will provide users a continuous presence view of up to four conferees at one time. Although the concept of the QCIF combiner is simple, many implementation issues need to be studied in more detail. The issues that need to be answered include the combiner architecture, required size of buffers, end-to-end delay, picture quality, and system complexity.

Two examples of continuous presence applications are shown in Fig. 1. In Fig. 1(a), conferees are involved in a multi-way videoconference. QCIF videos from the conferees are transmitted to the MCU with a transmission rate R. For each conferee, the MCU combines the selected four QCIF videos into a CIF video and transmits it back to the conferee at a transmission rate of 4R. Fig. 1(b) is a remote classroom application where the MCU combines 4 QCIF remote-site videos into a CIF video so that the teacher can interact with 4 remote sites simultaneously. The video from the teacher is broadcast to all the remote students.

The organization of this paper is as follows. Section II describes and compares pixel-domain and coded-domain combining. The remaining sections are focused on the coded-domain QCIF combiner. Section III describes the architectures of the QCIF combiner. Section IV presents a theoretical analysis for the delay and buffer size. Section V shows some simulation results. Section VI discusses techniques to improve the end-to-end delay. Finally, conclusions are provided in Section VII.

II. Coded-Domain vs. Pel-Domain Video Combining

Since the raw video data rate is large and the network bandwidth is limited, video data usually have to be compressed for network transmission. Therefore, video bridging in the network would normally require video decoding, combining in the pel domain, and encoding for retransmission. For some coded bit streams, however, video combining can be done in the coded domain. We define coded-domain video combining as a process that does not decode the compressed data down to the pel domain. The compressed data are either not decoded at all or only partially decoded for combining. In the undecoded case, e.g., the proposed QCIF combiner, the video bridge only has to process data headers and concatenate the remaining data stream without modification. In this case, we are mainly dealing with data coded as the variable-length codes (VLCs). We will refer to this case by *VLC-domain approach*. In the partially decoded case, video bridging is usually done after decoding variable-length codes, e.g., DCT-domain video compositing [5]. For pel domain video combining, the compressed data have to be fully decoded and then, after combining in the pel domain, encoded again. Table I gives a qualitative summary of the pros and cons of these

Manuscript received June 22, 1993; revised November 5, 1993 and January 26, 1994. This paper was recommended by King N. Ngan.
The authors are with Bellcore, Red Bank, NJ 07701-5699 USA.
IEEE Log Number 9404692.

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Mixing MCU

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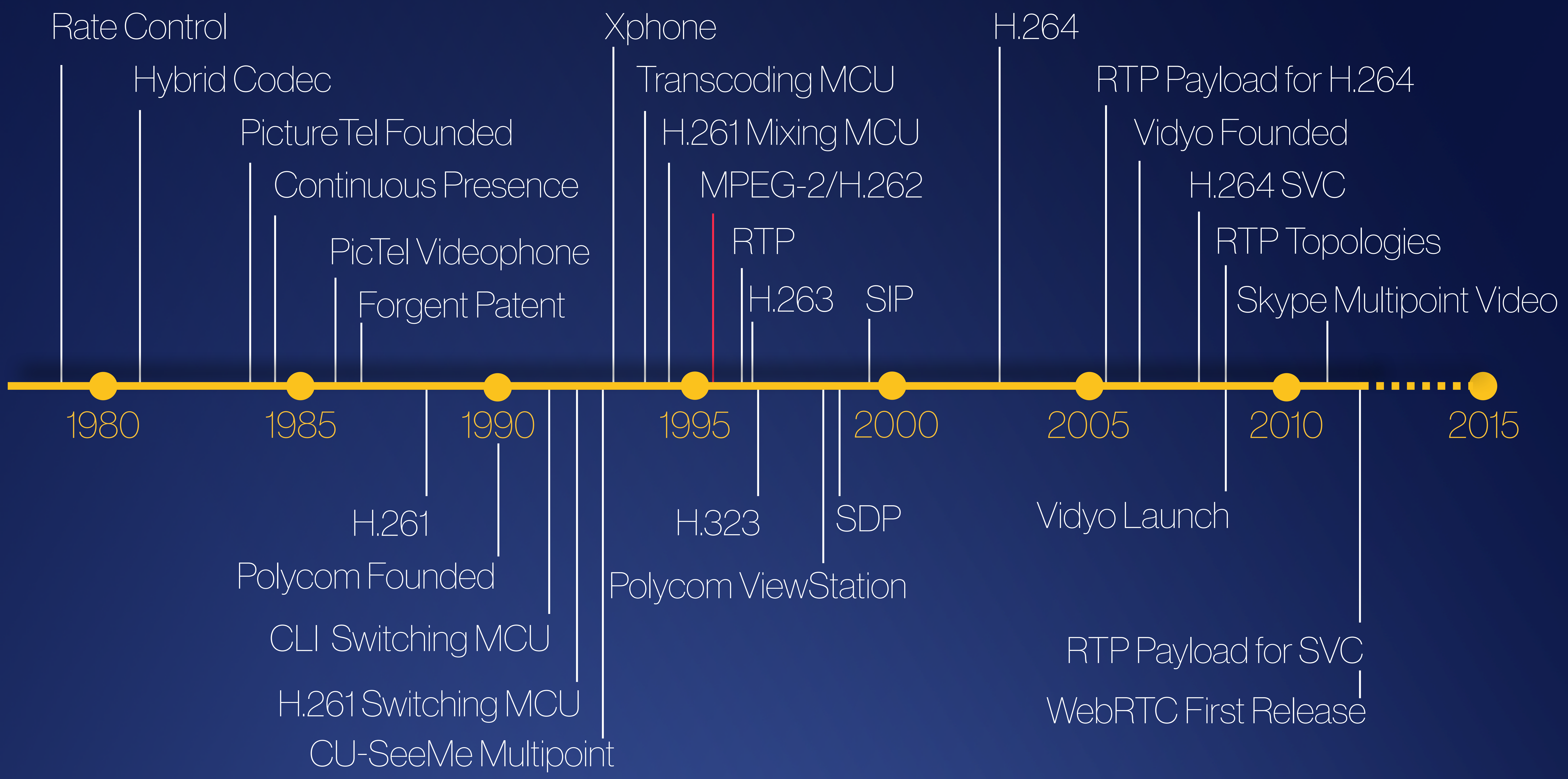

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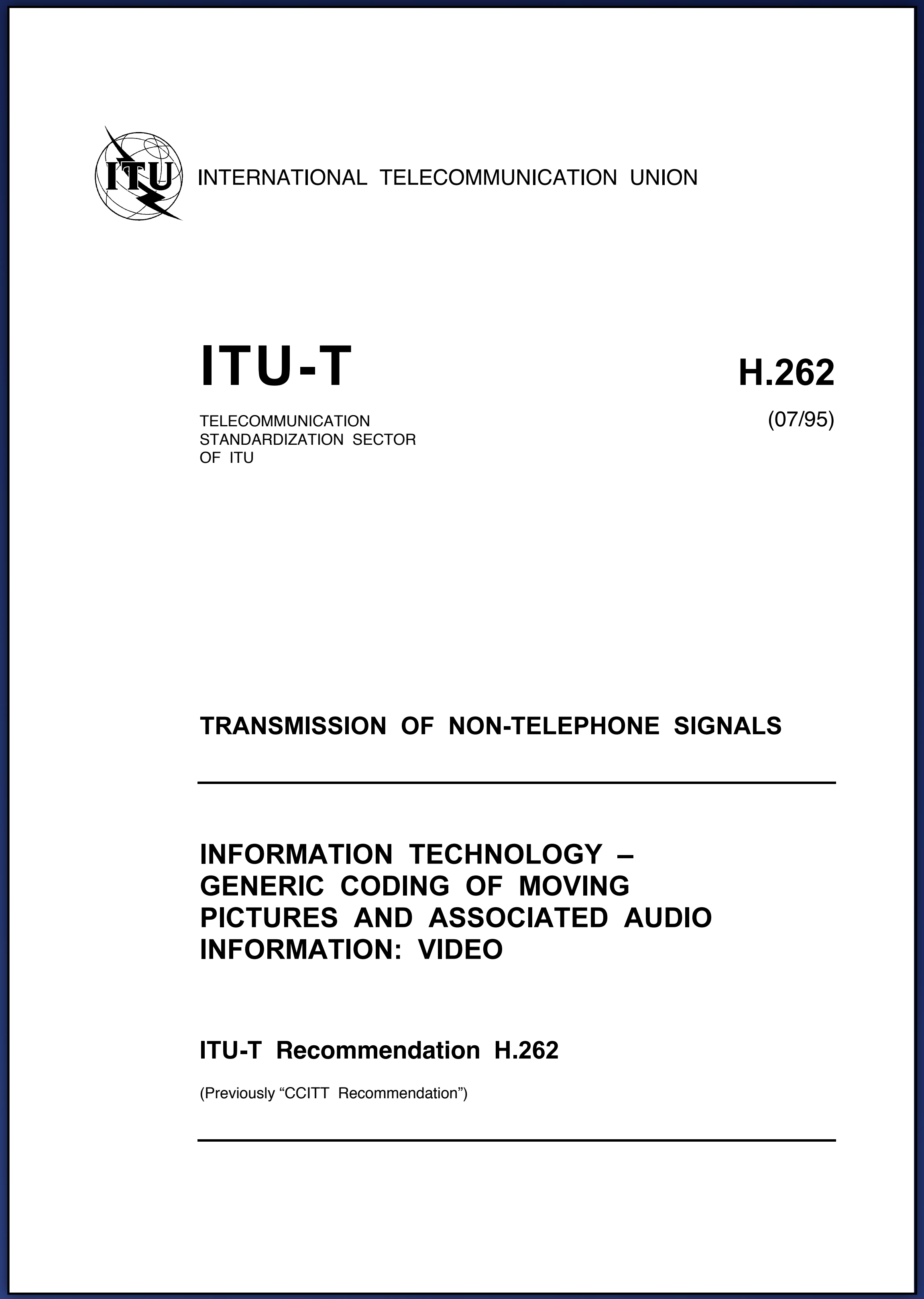
MPEG-2/H.262

July 1995

First digital video compression standard for broadcast, cable, and satellite TV, and DVD


INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

H.262

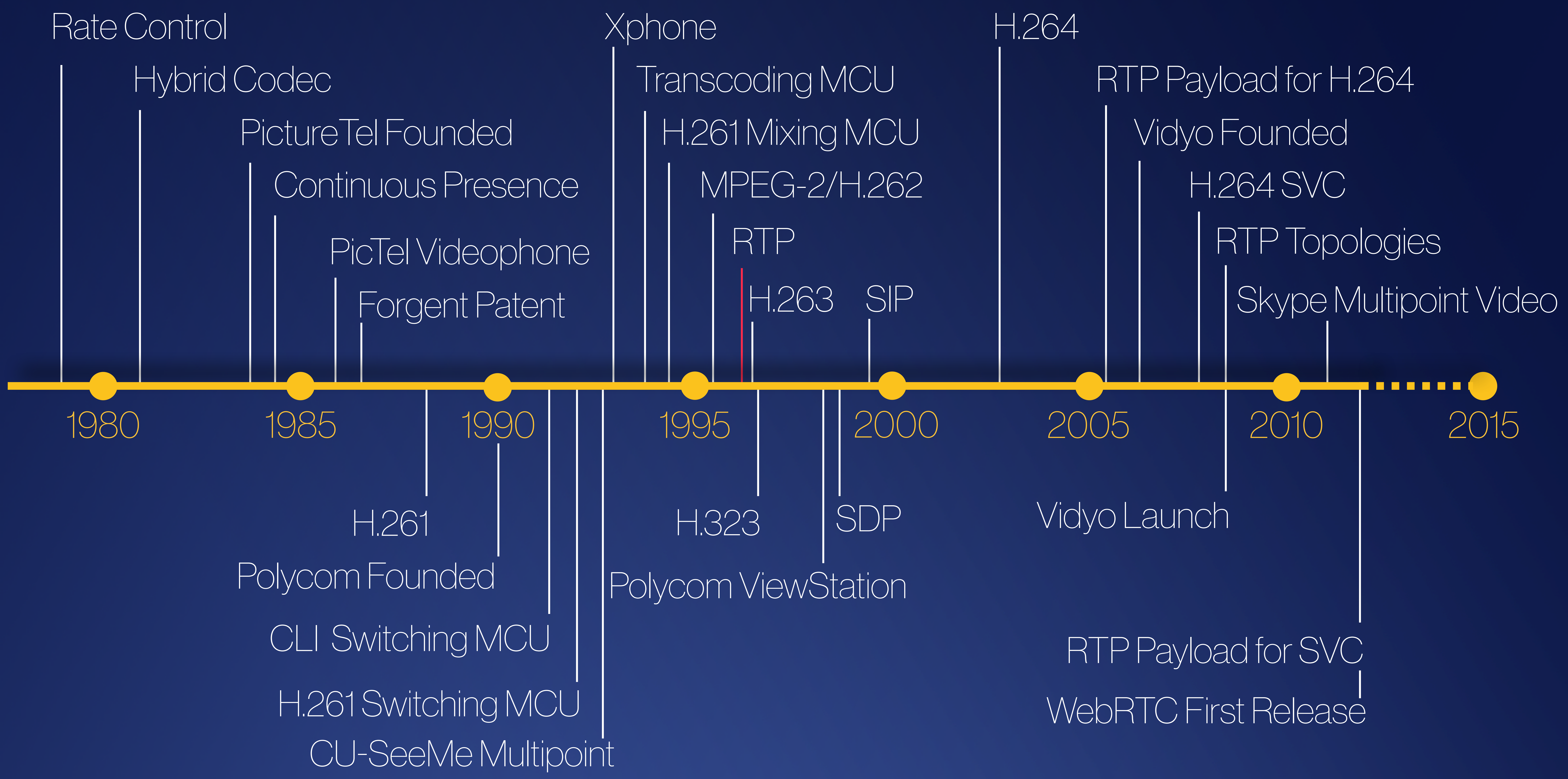
(07/95)

TRANSMISSION OF NON-TELEPHONE SIGNALS

INFORMATION TECHNOLOGY – GENERIC CODING OF MOVING PICTURES AND ASSOCIATED AUDIO INFORMATION: VIDEO

ITU-T Recommendation H.262

(Previously "CCITT Recommendation")



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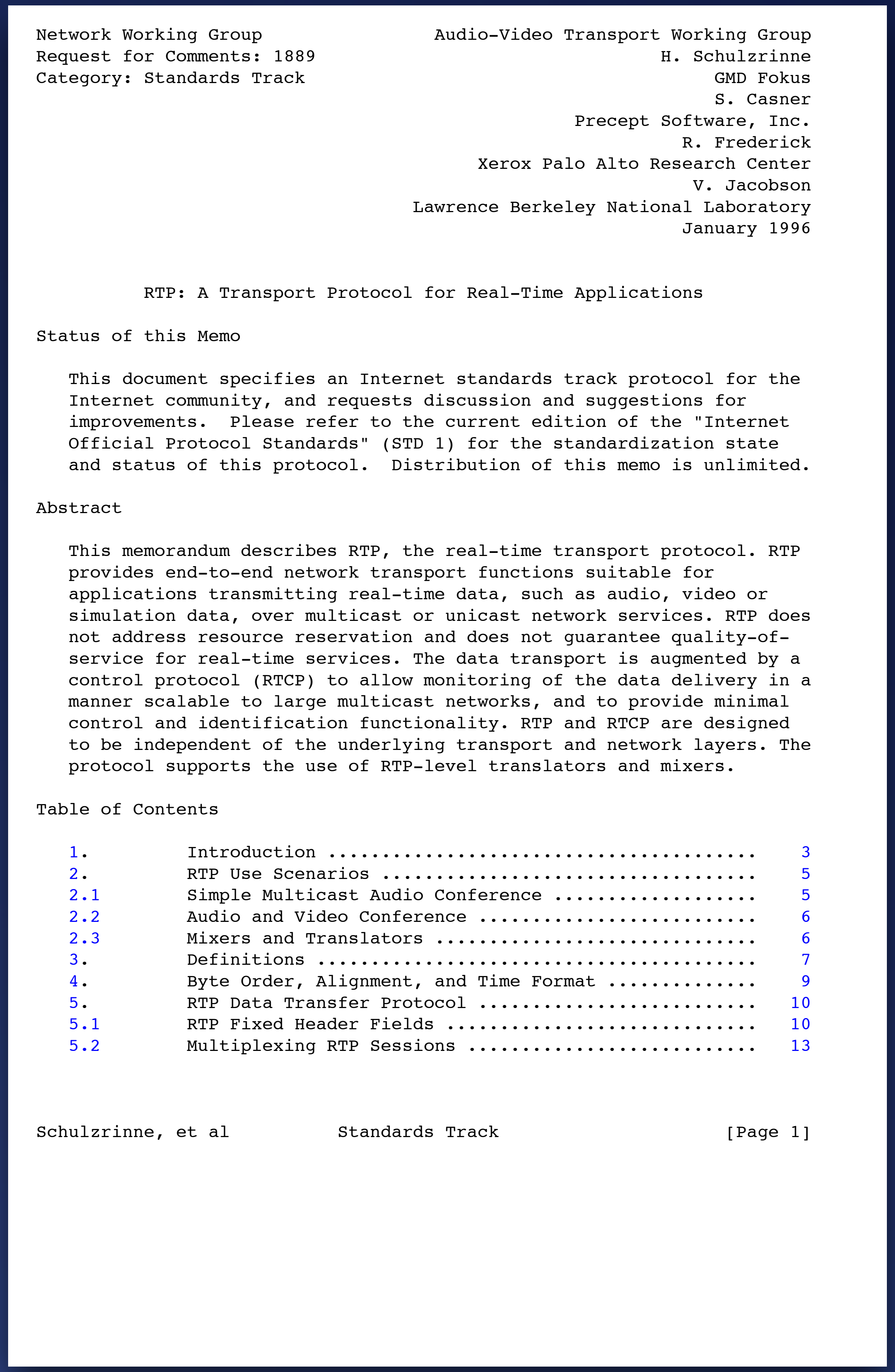
RTP

January 1996
Schulzrinne, Casner, Frederick, and Jacobson
RFC 1889

Original RTP version



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Network Working Group                 Audio-Video Transport Working Group
Request for Comments: 1889                                 H. Schulzrinne
Category: Standards Track                                       GMD Fokus
                                                                S. Casner
                                                    Precept Software, Inc.
                                                             R. Frederick
                                              Xerox Palo Alto Research Center
                                                              V. Jacobson
                                        Lawrence Berkeley National Laboratory
                                                             January 1996


          RTP: A Transport Protocol for Real-Time Applications

Status of this Memo

   This document specifies an Internet standards track protocol for the
   Internet community, and requests discussion and suggestions for
   improvements.  Please refer to the current edition of the "Internet
   Official Protocol Standards" (STD 1) for the standardization state
   and status of this protocol.  Distribution of this memo is unlimited.

Abstract

   This memorandum describes RTP, the real-time transport protocol. RTP
   provides end-to-end network transport functions suitable for
   applications transmitting real-time data, such as audio, video or
   simulation data, over multicast or unicast network services. RTP does
   not address resource reservation and does not guarantee quality-of-
   service for real-time services. The data transport is augmented by a
   control protocol (RTCP) to allow monitoring of the data delivery in a
   manner scalable to large multicast networks, and to provide minimal
   control and identification functionality. RTP and RTCP are designed
   to be independent of the underlying transport and network layers. The
   protocol supports the use of RTP-level translators and mixers.

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Schulzrinne, et al          Standards Track                     [Page 1]
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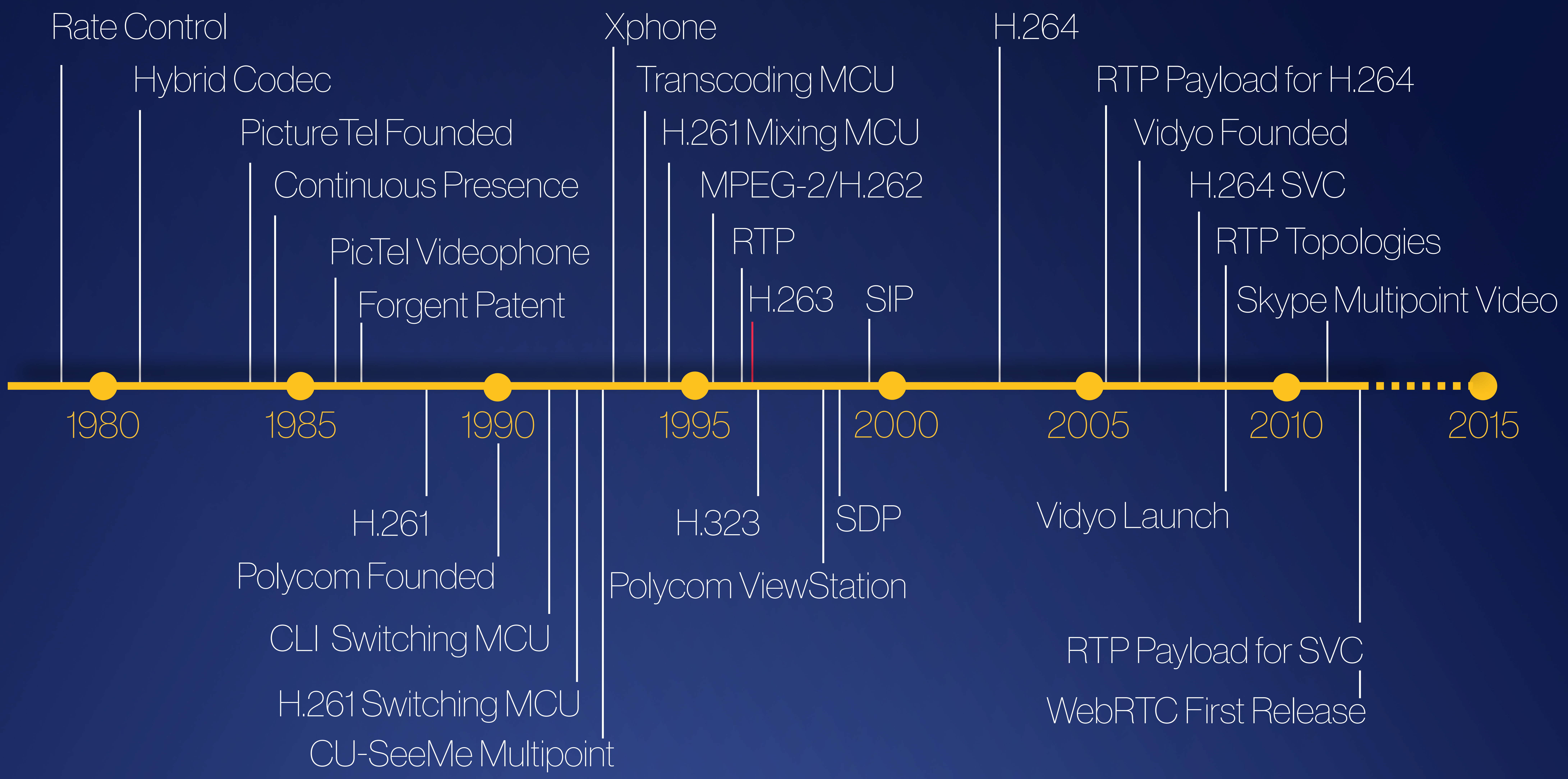


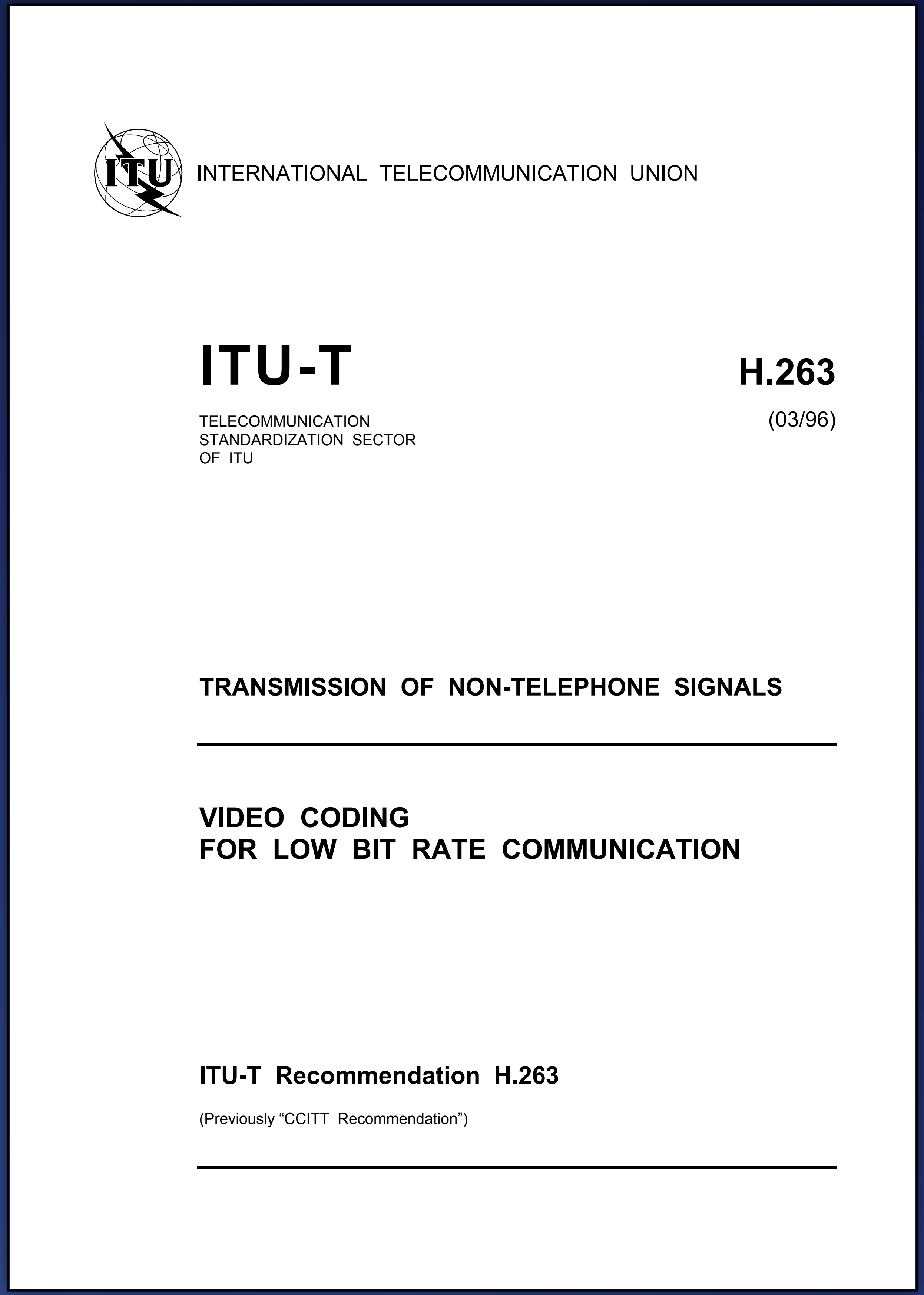
Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

H.263

March 1996

2nd generation video compression standard for communications



INTERNATIONAL TELECOMMUNICATION UNION

ITU-T — TELECOMMUNICATION STANDARDIZATION SECTOR OF ITU

H.263 (03/96)

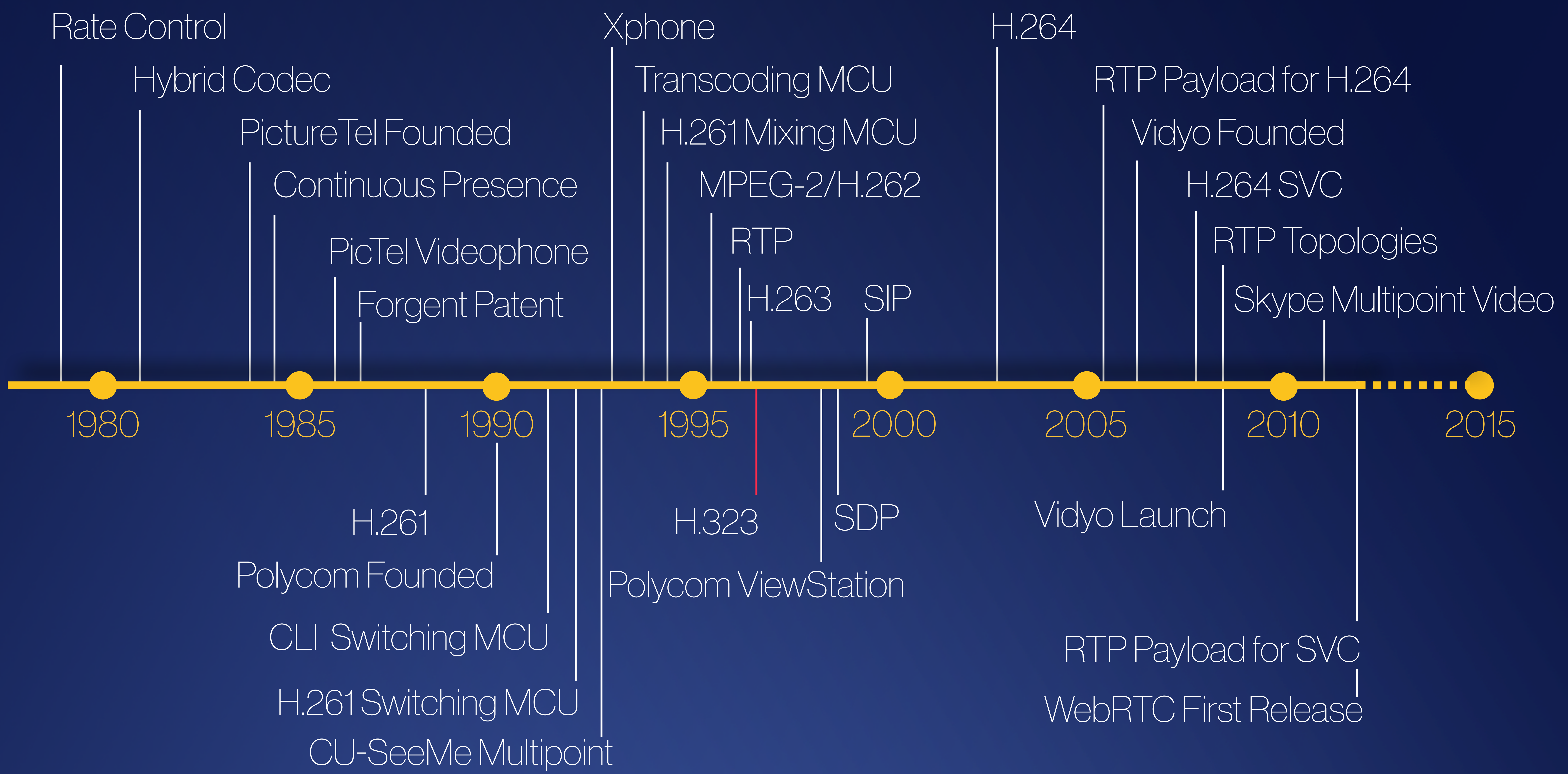
TRANSMISSION OF NON-TELEPHONE SIGNALS

VIDEO CODING FOR LOW BIT RATE COMMUNICATION

ITU-T Recommendation H.263

(Previously "CCITT Recommendation")

Vidyo®
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

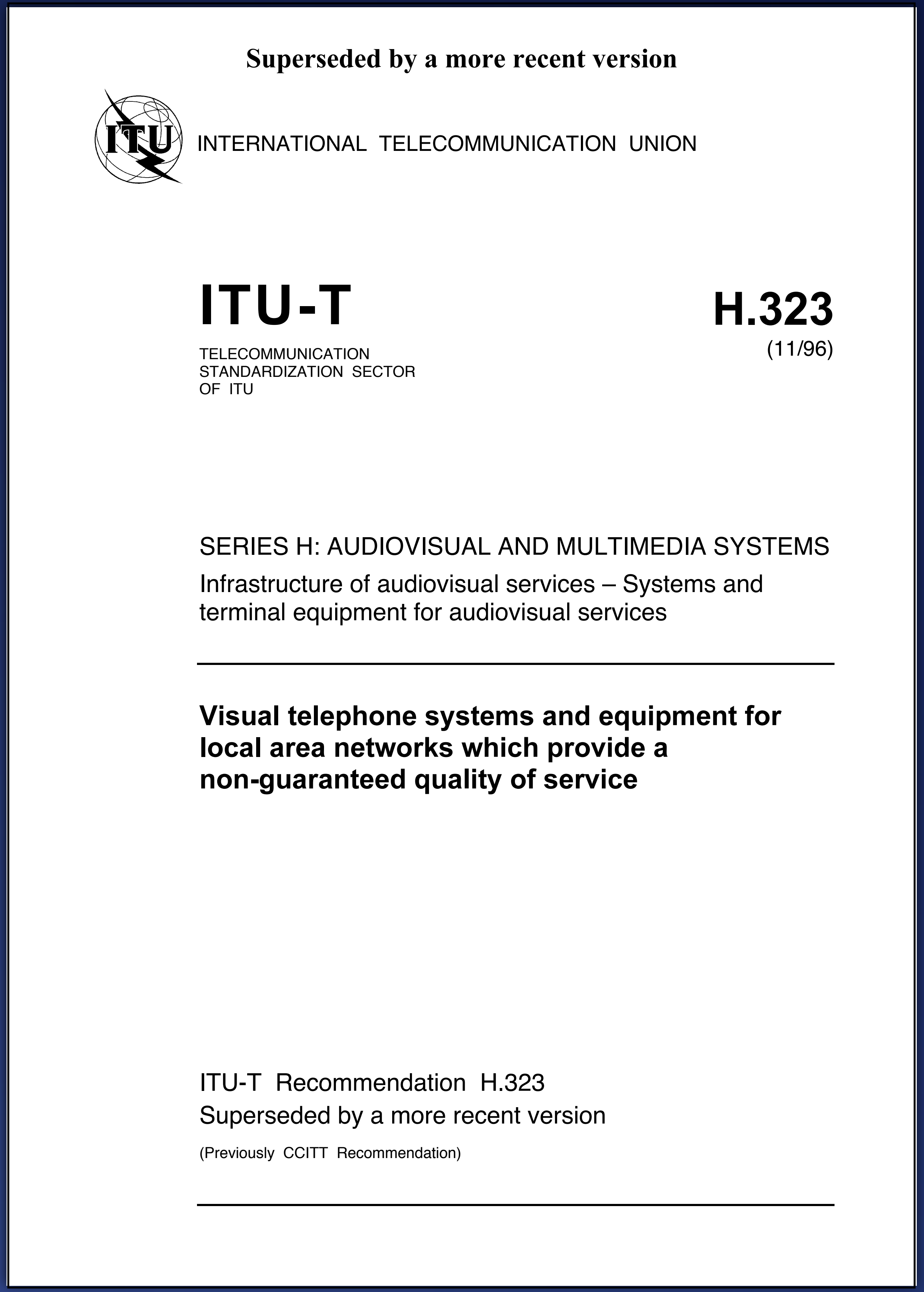


Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

H.323

November 1996

Packet-based multimedia communication systems


Superseded by a more recent version

INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

H.323
(11/96)

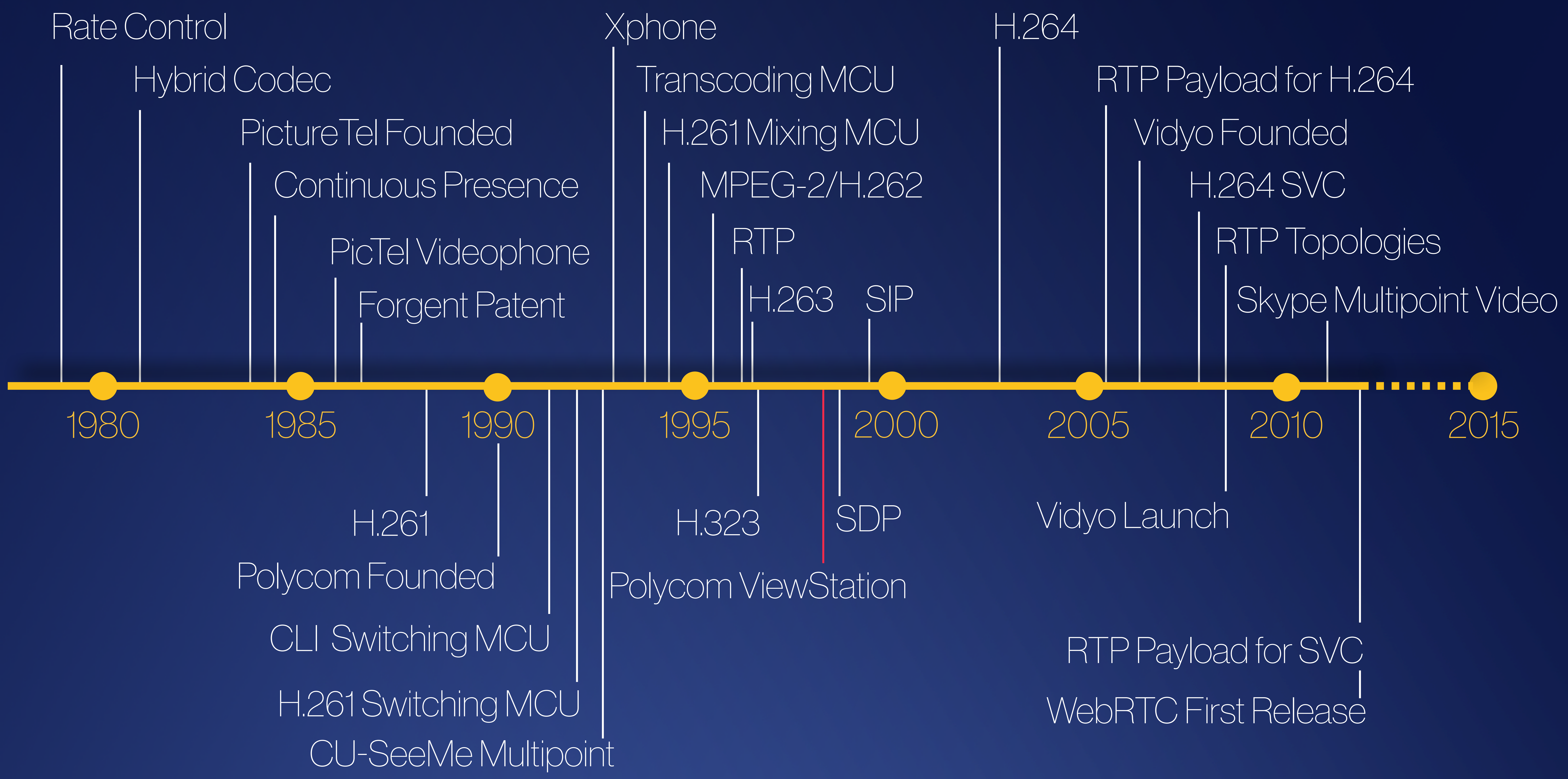
SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS
Infrastructure of audiovisual services – Systems and terminal equipment for audiovisual services

Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service

ITU-T Recommendation H.323
Superseded by a more recent version

(Previously CCITT Recommendation)

Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

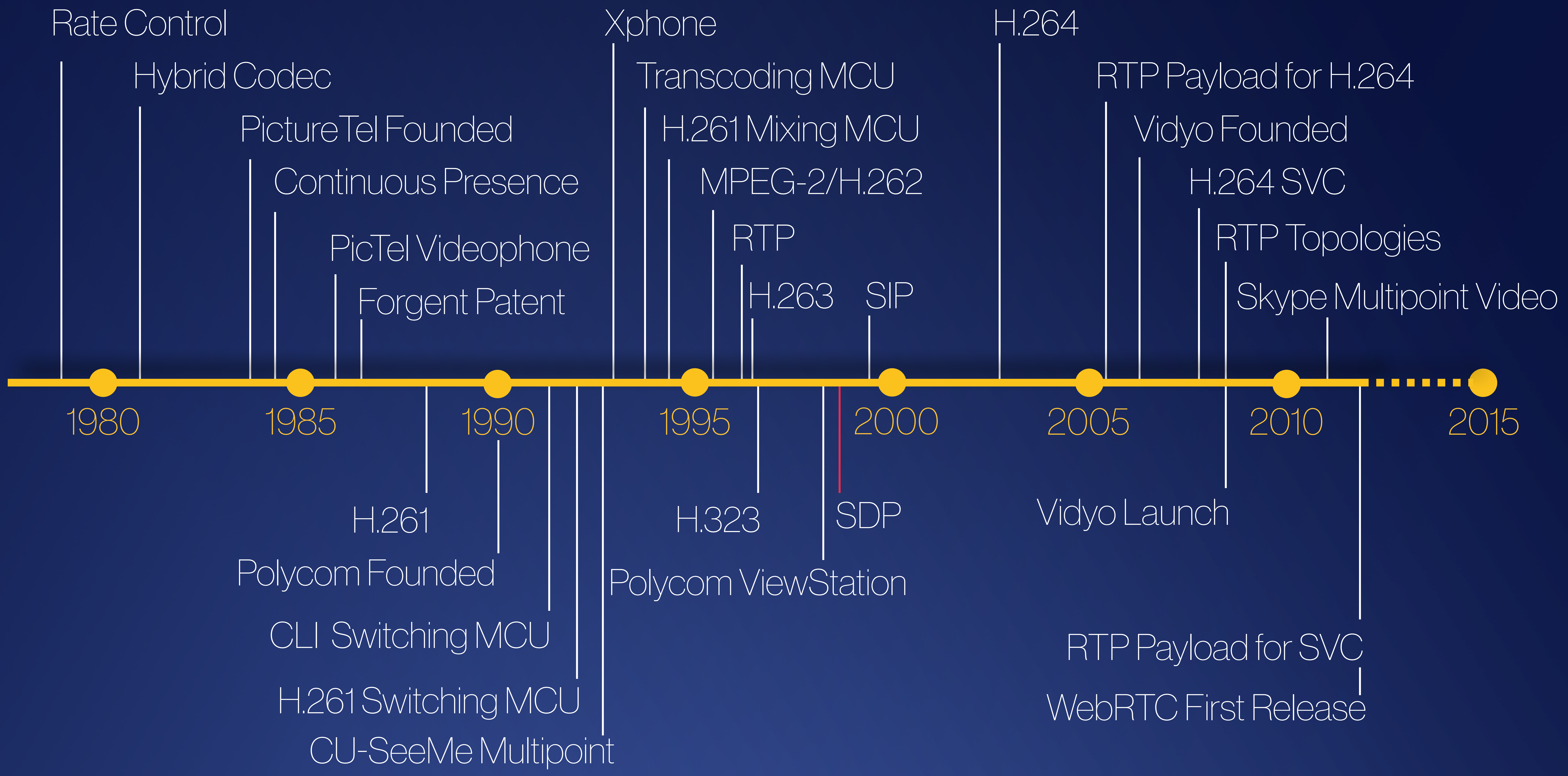


Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Polycom ViewStation

January 1998
First Polycom video product

H.323 over IP
H.261 and H.263 video
4-way embedded multipoint
(auto, discussion/continuous
presence, presentation, active
speaker switching)



Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

SDP

April 1998

Handley and Jacobson

RFC 2327

Session Description Protocol

Network Working Group — M. Handley
Request for Comments: 2327 — V. Jacobson
Category: Standards Track — ISI/LBNL
April 1998

SDP: Session Description Protocol

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

Copyright Notice

Abstract

This document defines the Session Description Protocol, SDP. SDP is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.

This document is a product of the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force. Comments are solicited and should be addressed to the working group's mailing list at confctrl@isi.edu and/or the authors.

1. Introduction

On the Internet multicast backbone (Mbone), a session directory tool is used to advertise multimedia conferences and communicate the conference addresses and conference tool-specific information necessary for participation. This document defines a session description protocol for this purpose, and for general real-time multimedia session description purposes. This memo does not describe multicast address allocation or the distribution of SDP messages in detail. These are described in accompanying memos. SDP is not intended for negotiation of media encodings.

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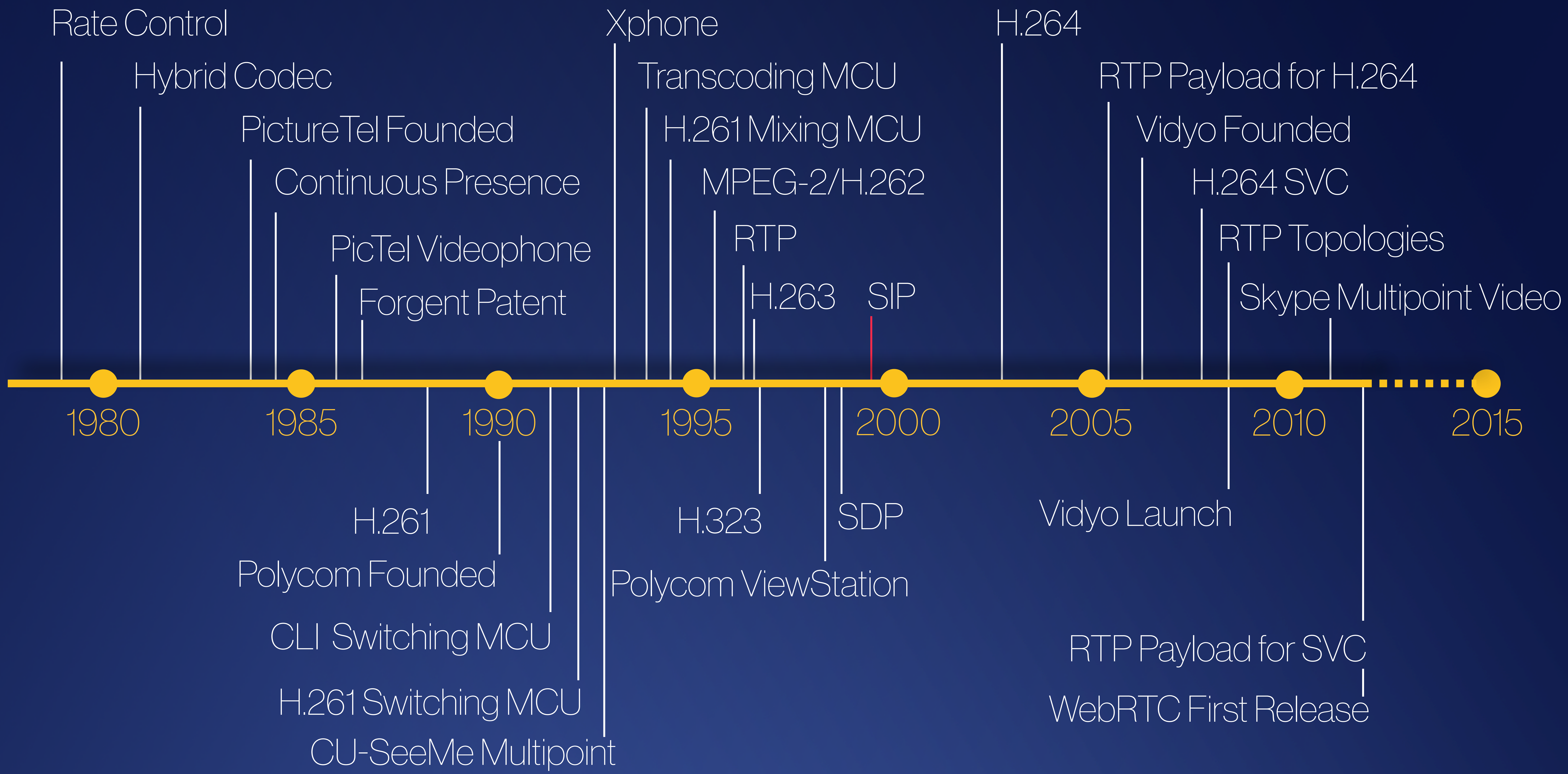
Chief Scientist & Co-founder



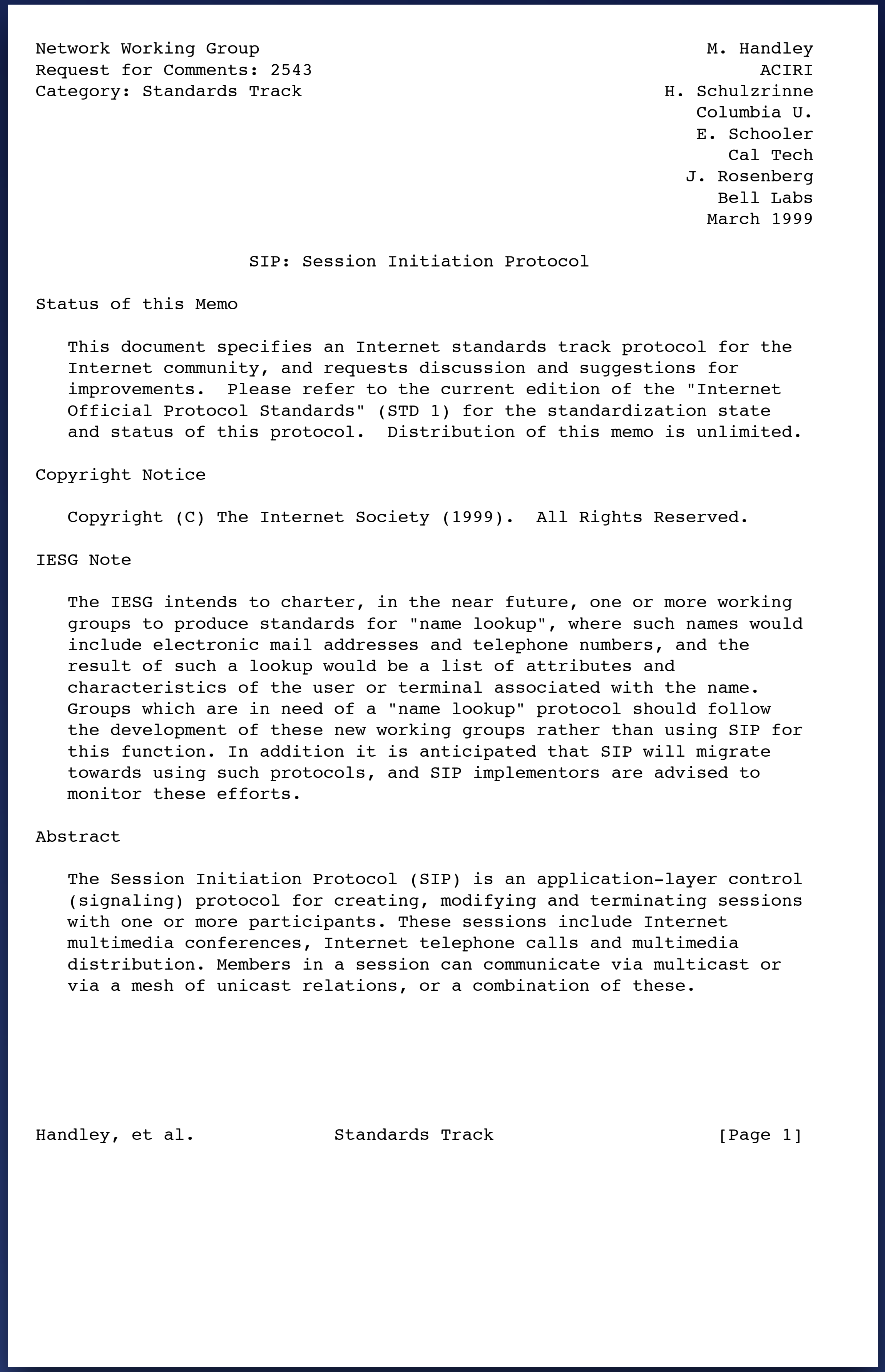
Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

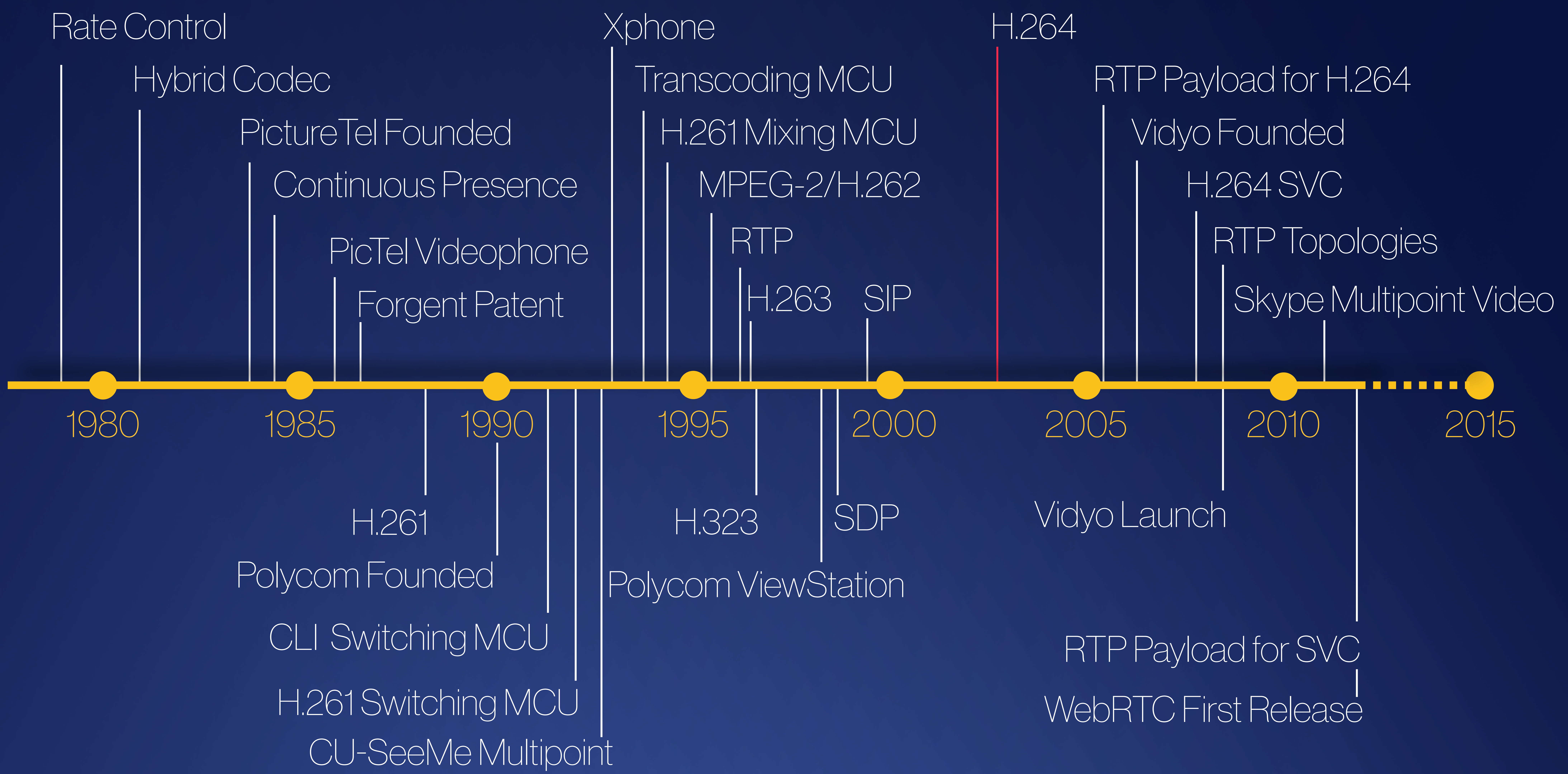
SIP

March 1999
Schulzrinne, Handley, Schooler, and Rosenberg
RFC 2543

Session Initiation Protocol



```
Network Working Group                                         M. Handley
Request for Comments: 2543                                         ACIRI
Category: Standards Track                                 H. Schulzrinne
                                                             Columbia U.
                                                              E. Schooler
                                                                Cal Tech
                                                             J. Rosenberg
                                                               Bell Labs
                                                              March 1999

                   SIP: Session Initiation Protocol

Status of this Memo

   This document specifies an Internet standards track protocol for the
   Internet community, and requests discussion and suggestions for
   improvements.  Please refer to the current edition of the "Internet
   Official Protocol Standards" (STD 1) for the standardization state
   and status of this protocol.  Distribution of this memo is unlimited.

Copyright Notice

   Copyright (C) The Internet Society (1999).  All Rights Reserved.

IESG Note

   The IESG intends to charter, in the near future, one or more working
   groups to produce standards for "name lookup", where such names would
   include electronic mail addresses and telephone numbers, and the
   result of such a lookup would be a list of attributes and
   characteristics of the user or terminal associated with the name.
   Groups which are in need of a "name lookup" protocol should follow
   the development of these new working groups rather than using SIP for
   this function. In addition it is anticipated that SIP will migrate
   towards using such protocols, and SIP implementors are advised to
   monitor these efforts.

Abstract

   The Session Initiation Protocol (SIP) is an application-layer control
   (signaling) protocol for creating, modifying and terminating sessions
   with one or more participants. These sessions include Internet
   multimedia conferences, Internet telephone calls and multimedia
   distribution. Members in a session can communicate via multicast or
   via a mesh of unicast relations, or a combination of these.

Handley, et al.              Standards Track                    [Page 1]
```



Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

H.264

May 2003

H.264/MPEG-4 Part 10
AVC

The ubiquitous video codec


INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

H.264
(05/2003)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS

Infrastructure of audiovisual services – Coding of moving video

Advanced video coding for generic audiovisual services

ITU-T Recommendation H.264

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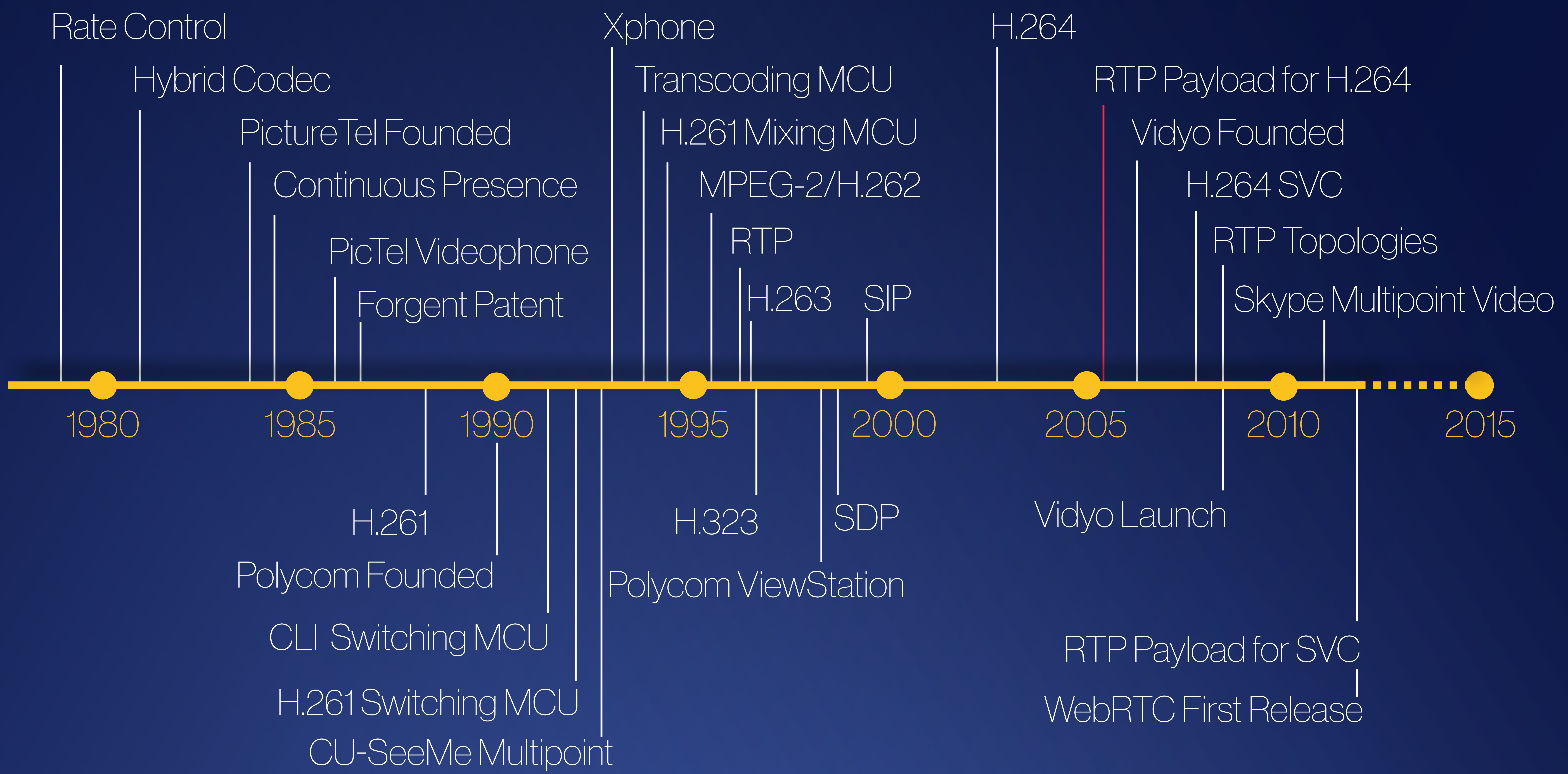
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

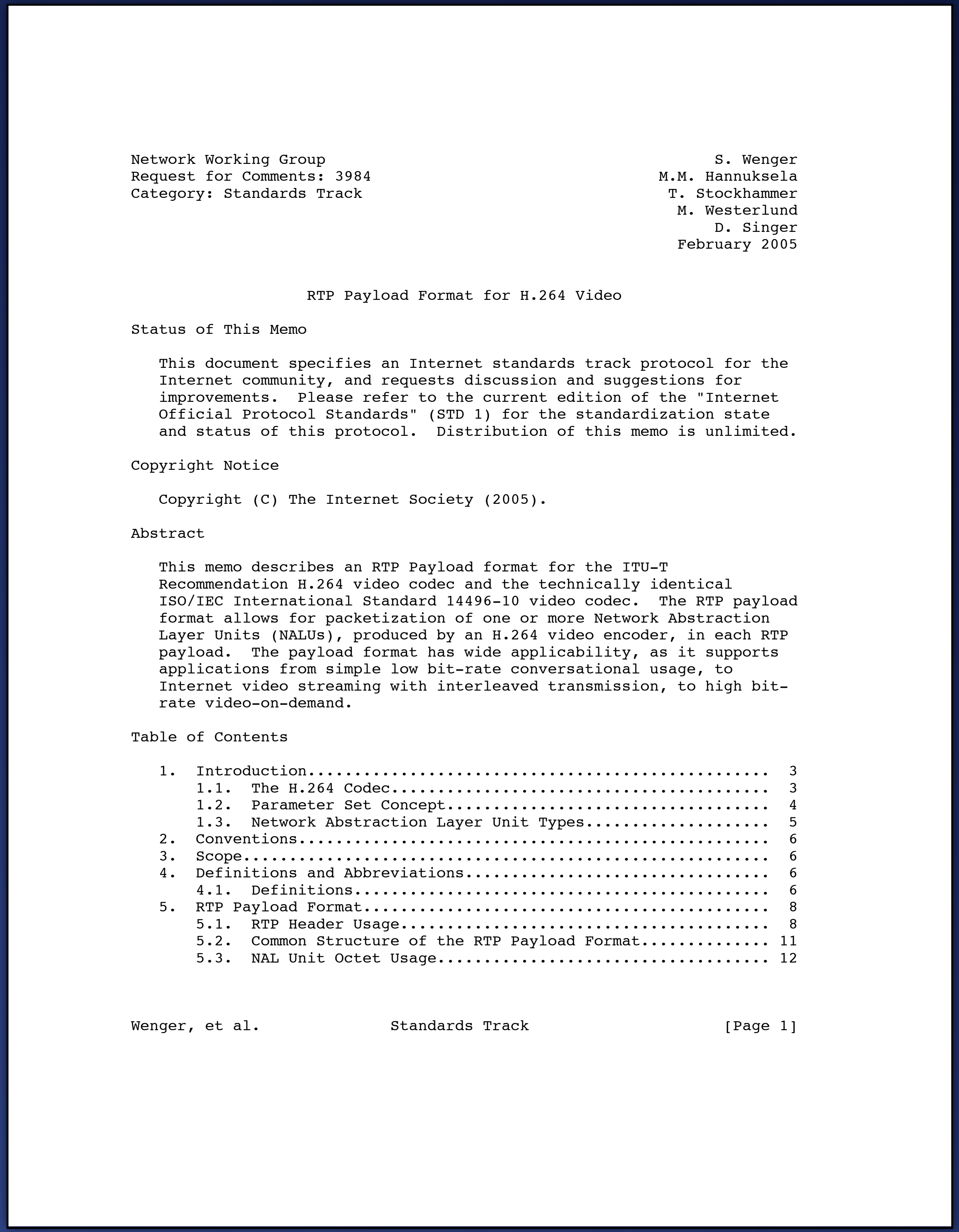


Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

RTP Payload for H.264

February 2005
Wenger, Stockhammer, Westerlund, and Singer
RFC 3984

How to packetize H.264 video for RTP transport, plus signaling parameters



```
Network Working Group                                          S. Wenger
Request for Comments: 3984                                 M.M. Hannuksela
Category: Standards Track                                  T. Stockhammer
                                                              M. Westerlund
                                                                  D. Singer
                                                              February 2005


                    RTP Payload Format for H.264 Video

Status of This Memo

   This document specifies an Internet standards track protocol for the
   Internet community, and requests discussion and suggestions for
   improvements.  Please refer to the current edition of the "Internet
   Official Protocol Standards" (STD 1) for the standardization state
   and status of this protocol.  Distribution of this memo is unlimited.

Copyright Notice

   Copyright (C) The Internet Society (2005).

Abstract

   This memo describes an RTP Payload format for the ITU-T
   Recommendation H.264 video codec and the technically identical
   ISO/IEC International Standard 14496-10 video codec.  The RTP payload
   format allows for packetization of one or more Network Abstraction
   Layer Units (NALUs), produced by an H.264 video encoder, in each RTP
   payload.  The payload format has wide applicability, as it supports
   applications from simple low bit-rate conversational usage, to
   Internet video streaming with interleaved transmission, to high bit-
   rate video-on-demand.

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   2.  Conventions.................................................   6
   3.  Scope.......................................................   6
   4.  Definitions and Abbreviations...............................   6
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Wenger, et al.              Standards Track                     [Page 1]
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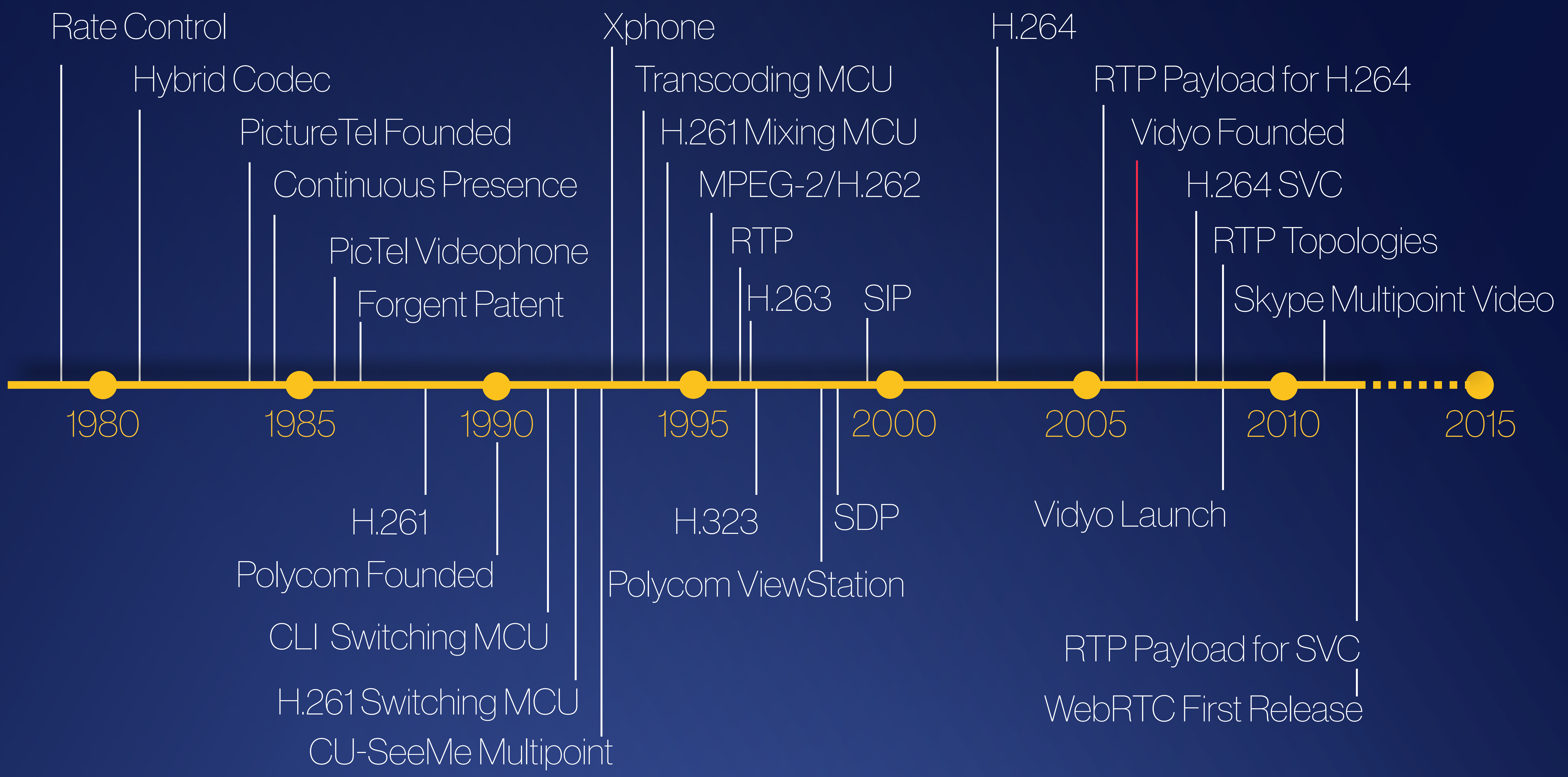

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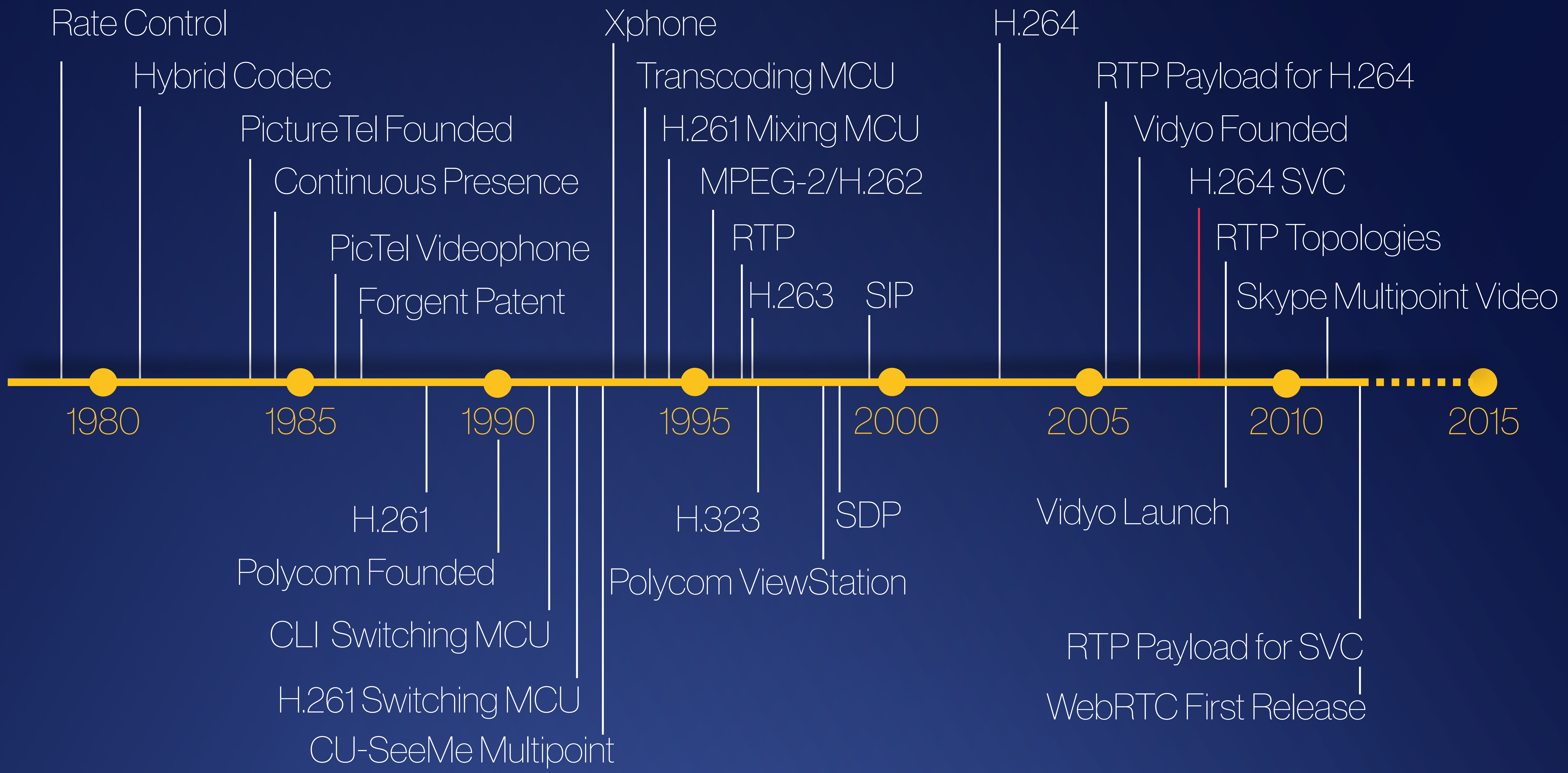
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Vidyo Founded

April 2005
(as "Layered Media")

Shapiro, More, and Eleftheriadis



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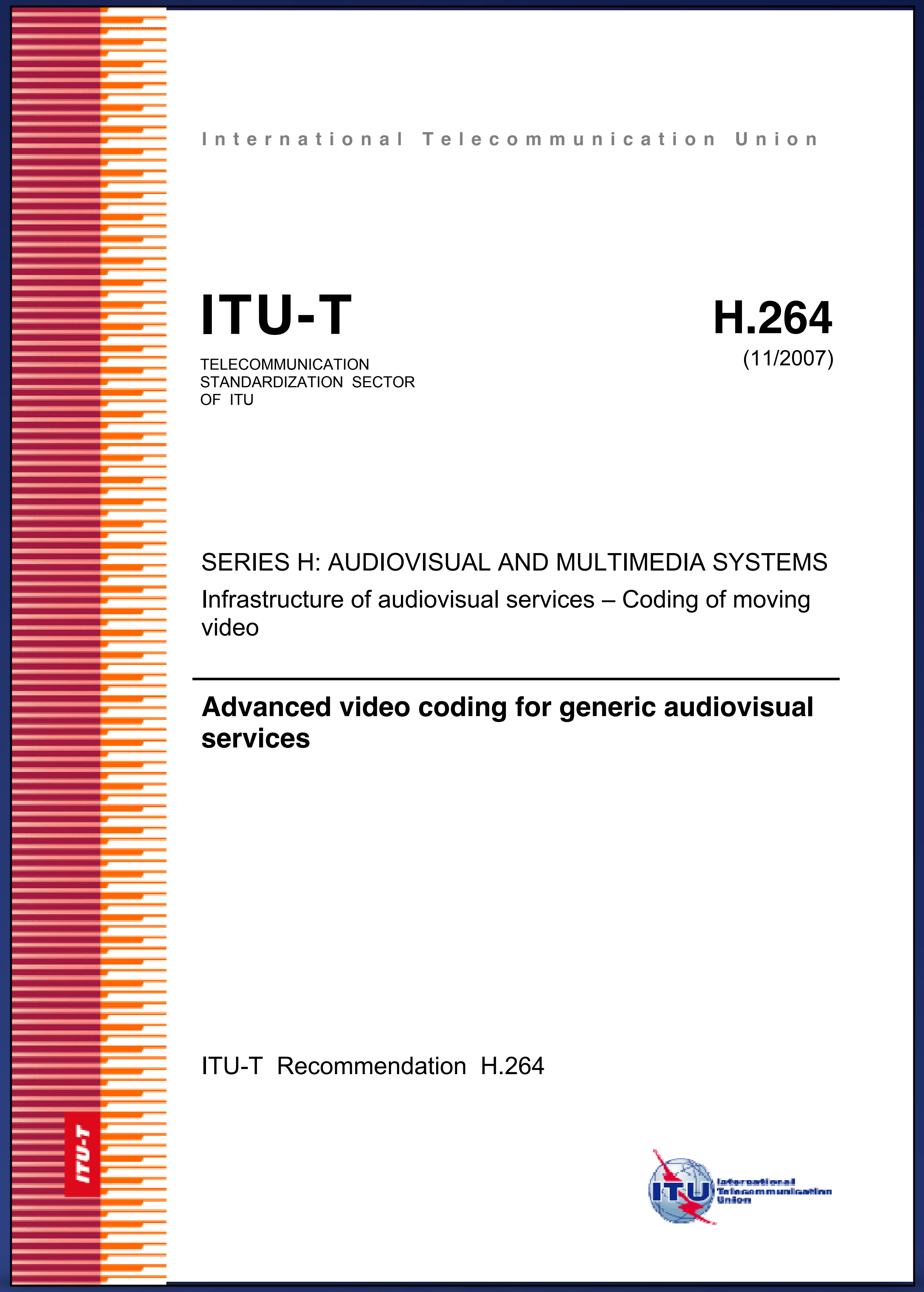
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

H.264 SVC

November 2007

H.264/MPEG-4 Part 10
AVC Annex G

Scalable Video Coding


International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

H.264
(11/2007)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS
Infrastructure of audiovisual services – Coding of moving video

Advanced video coding for generic audiovisual services

ITU-T Recommendation H.264

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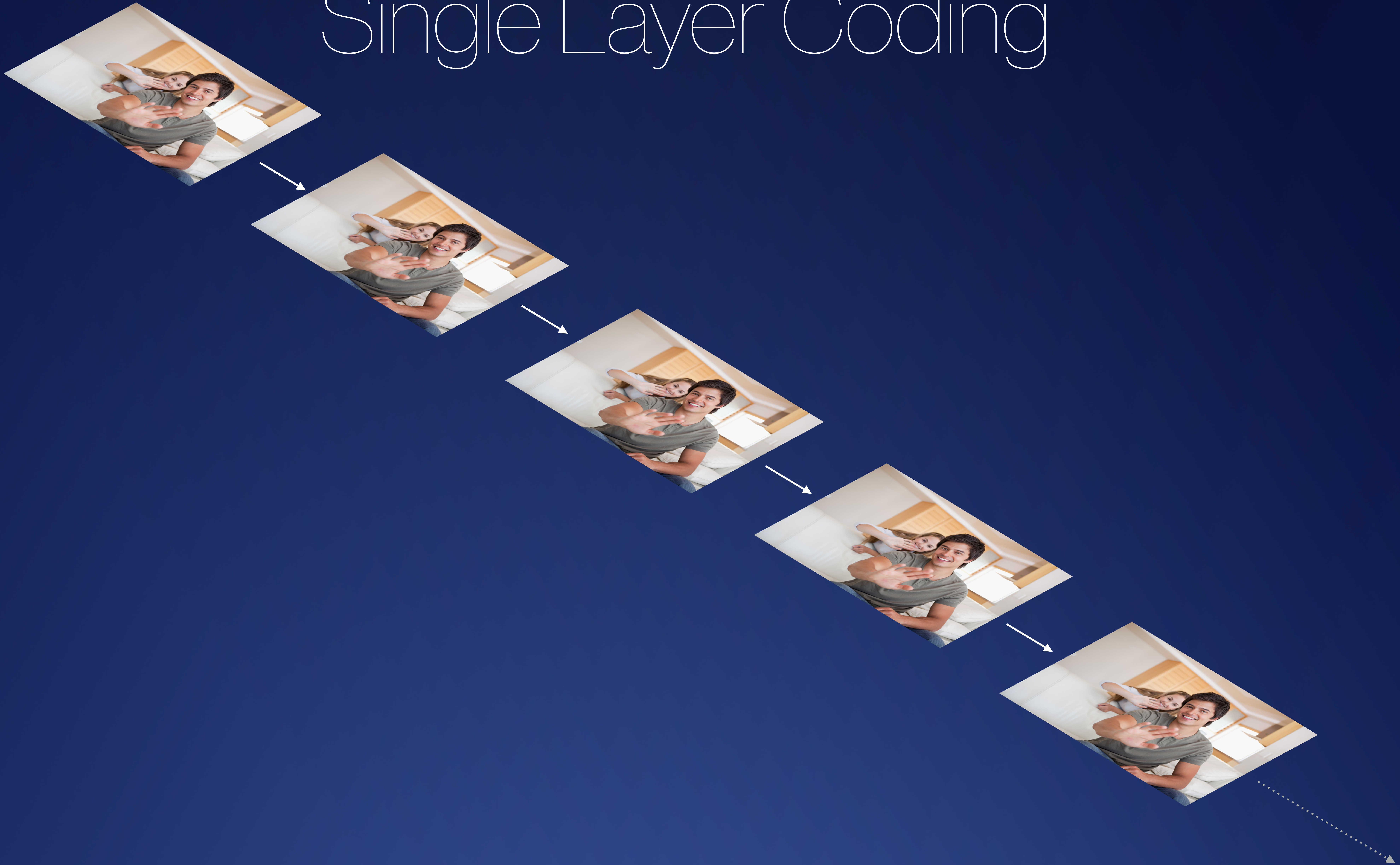
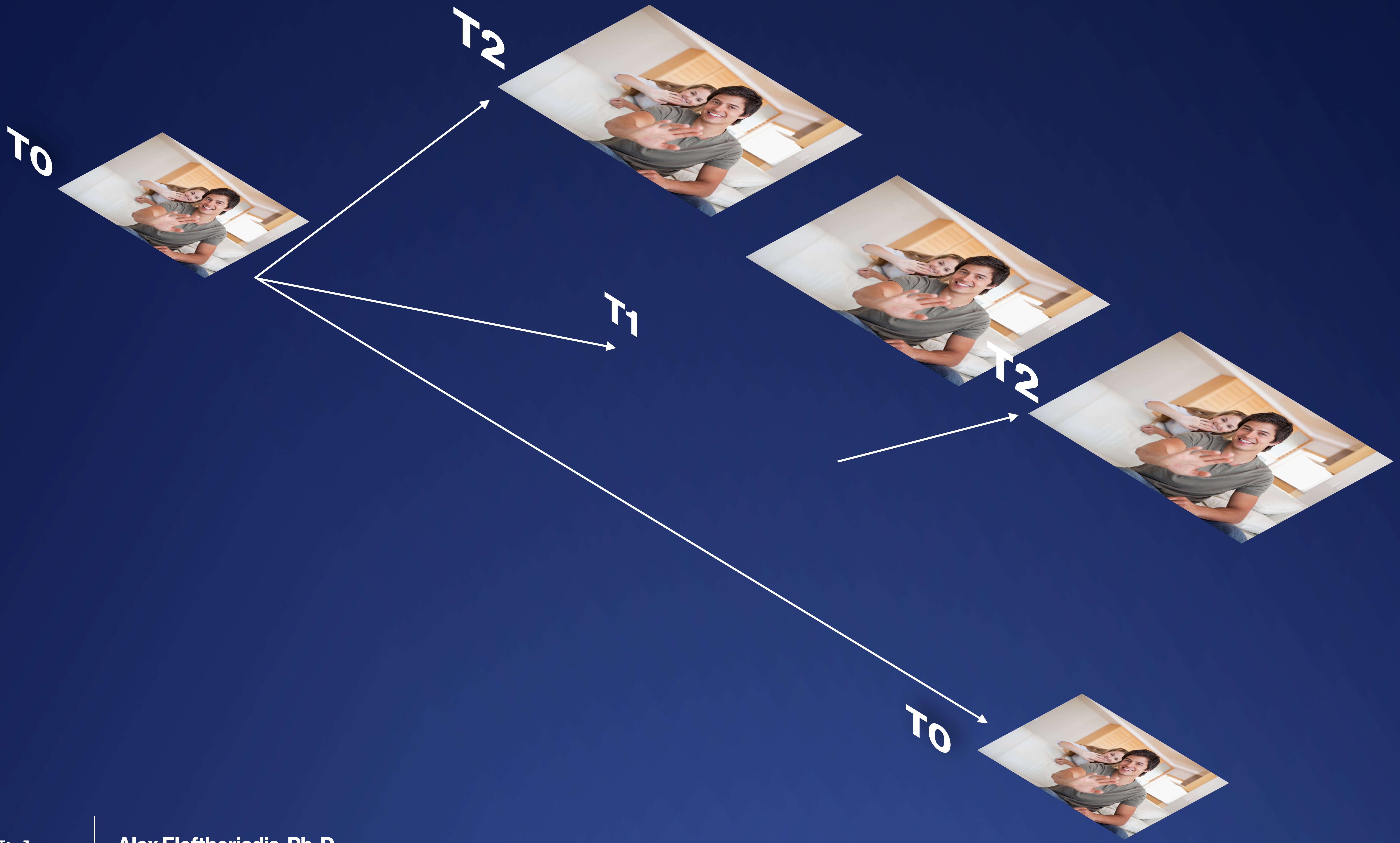
Chief Scientist & Co-founder

SVC

Single Layer Coding



Temporal Scalability



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Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Spatial Scalability

Spatial Scalability

Pros & Cons

Pros & Cons

~ 20% more bits than single layer

superb error resilience - can take more than 20% packet loss rate

adaptability

Scalability is already in:

H.264

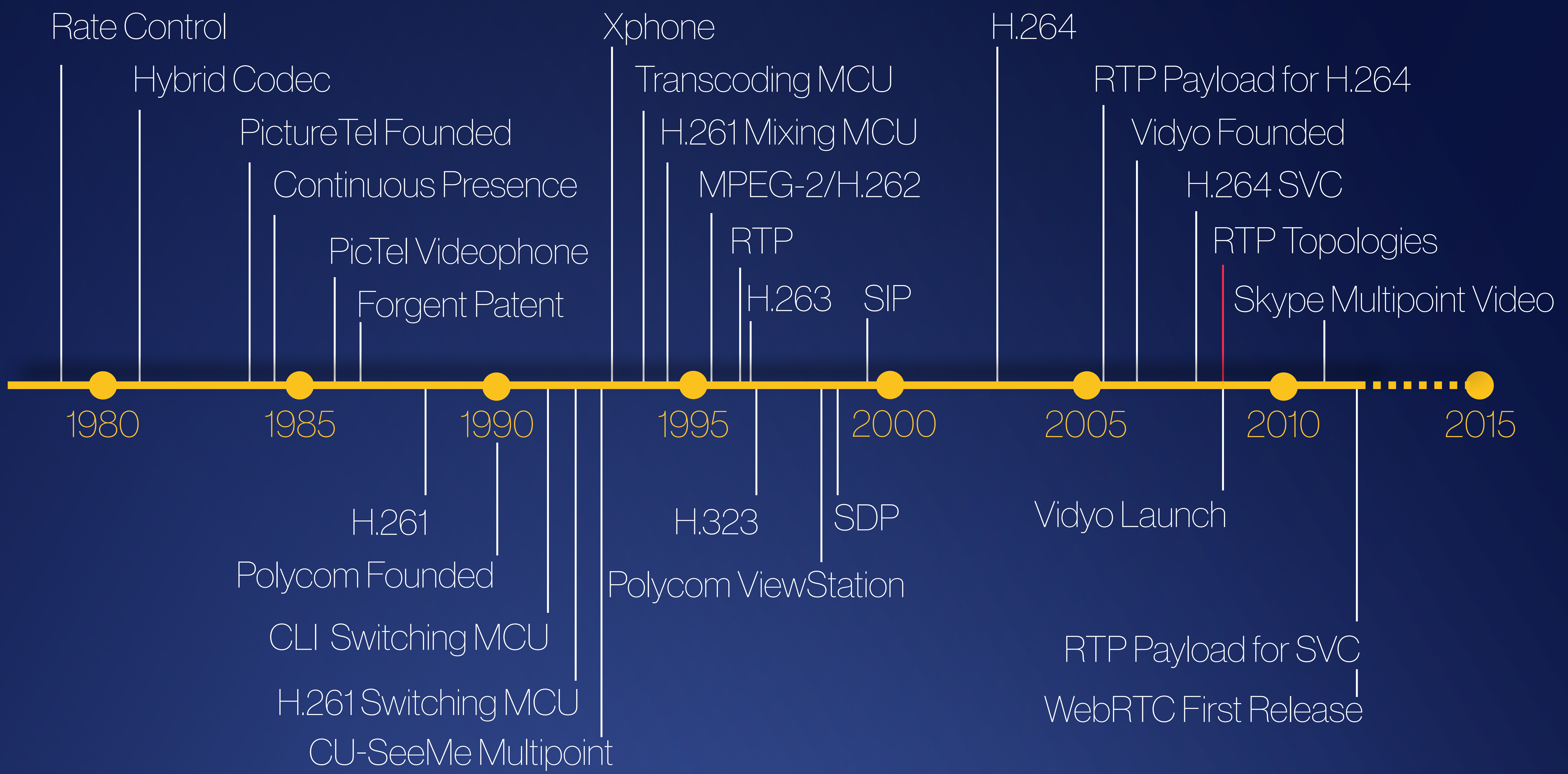
VP8

H.265/HEVC

H.265 v.2 - Scalable HEVC (October 2014)

VP9 (in progress - Vidyo/Google)

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

RTP Topologies

January 2008

Westerlund and Wenger

RFC 5117

Multipoint topologies used in RTP-based environments, particularly centralized topologies for videoconferencing

Network Working Group
Request for Comments: 5117
Category: Informational

M. Westerlund
Ericsson
S. Wenger
Nokia

January 2008

RTP Topologies

Status of This Memo

This memo provides information for the Internet community. It does not specify an Internet standard of any kind. Distribution of this memo is unlimited.

Abstract

This document discusses multi-endpoint topologies used in Real-time Transport Protocol (RTP)-based environments. In particular, centralized topologies commonly employed in the video conferencing industry are mapped to the RTP terminology.

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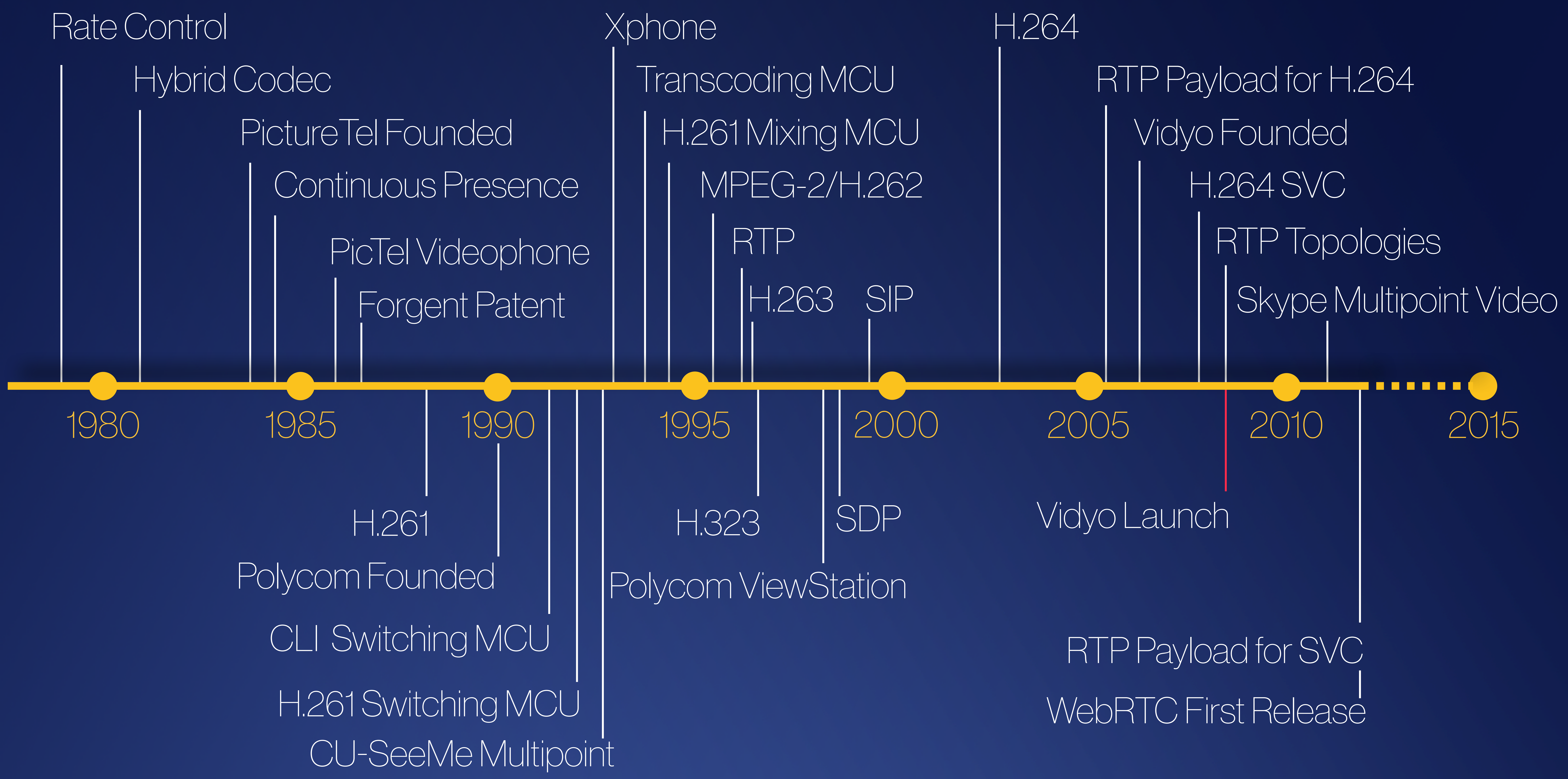
Chief Scientist & Co-founder



Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Vidyo Launch

January 2008

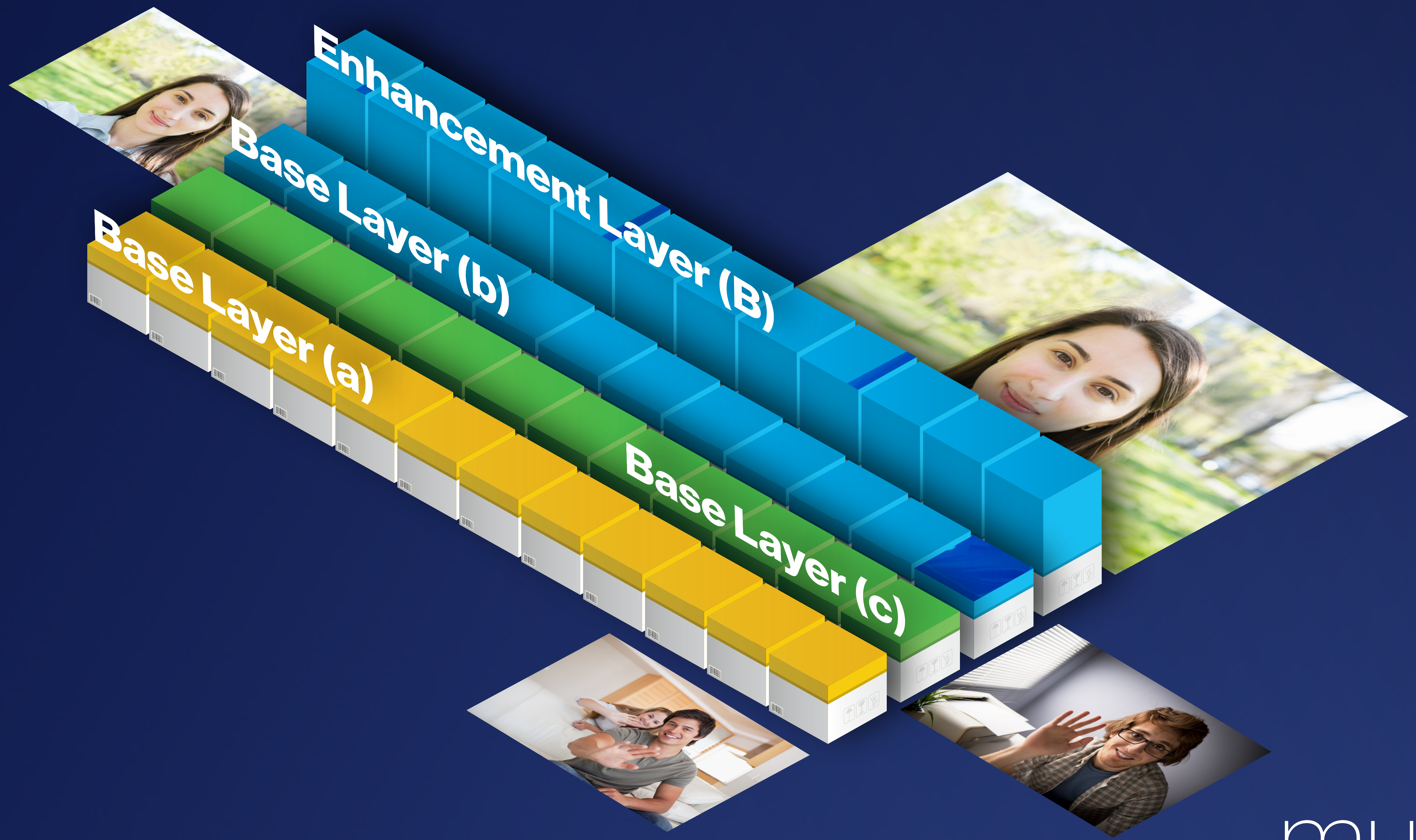
Vidyo launches SVC and the VidyoRouter® architecture

A new type of server - VidyoRouter

A new type of server - VidyoRouter

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Endpoint design



multi-stream
+
composition in endpoint
~=
browser

Perfect match for WebRTC

What did we gain?

What did we gain?

Error Resilience

Rate Matching

Personalized Layout

Low Delay

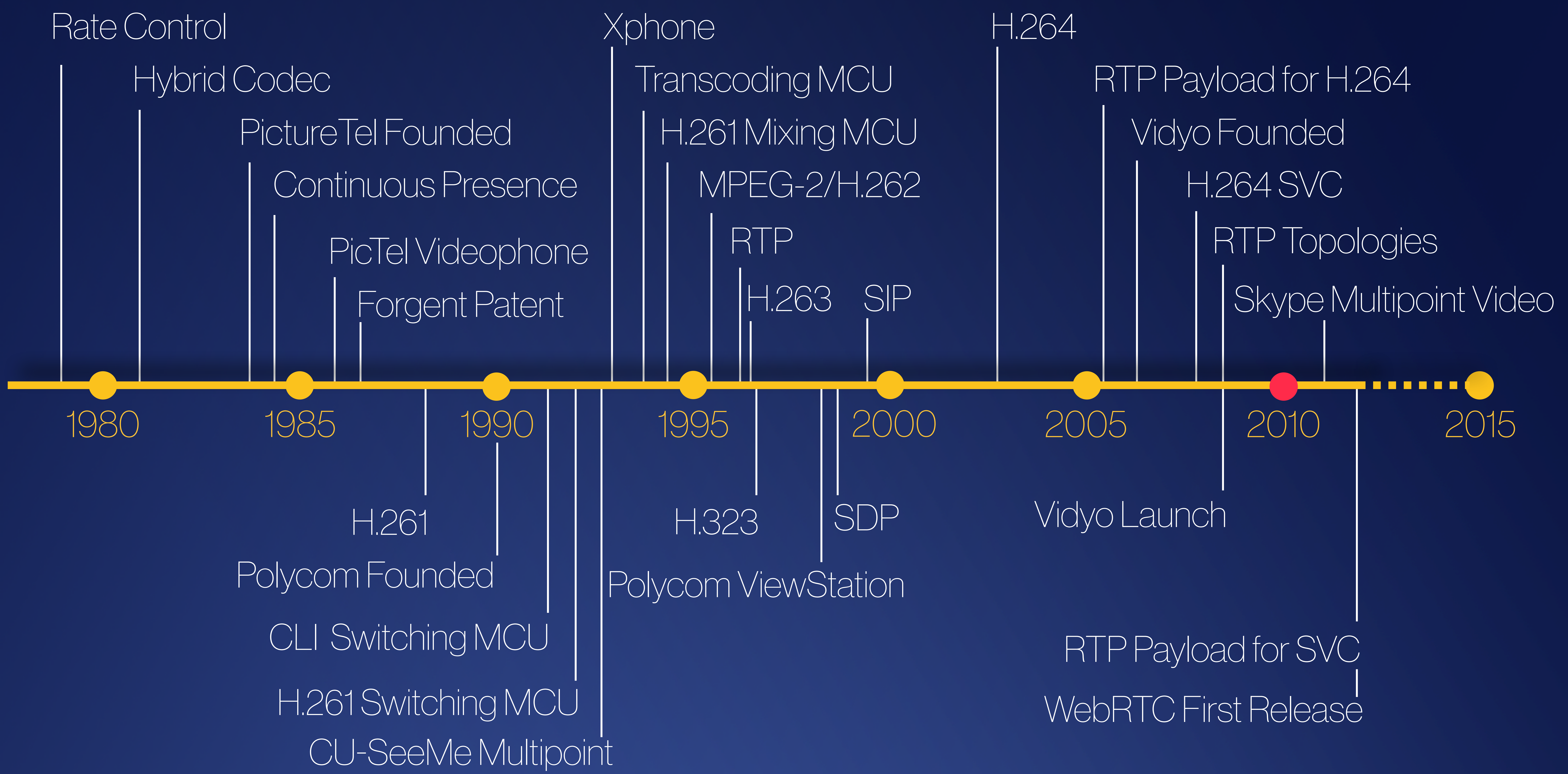
Error Localization

Low Complexity

Cascading

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

All with no signal processing at the server



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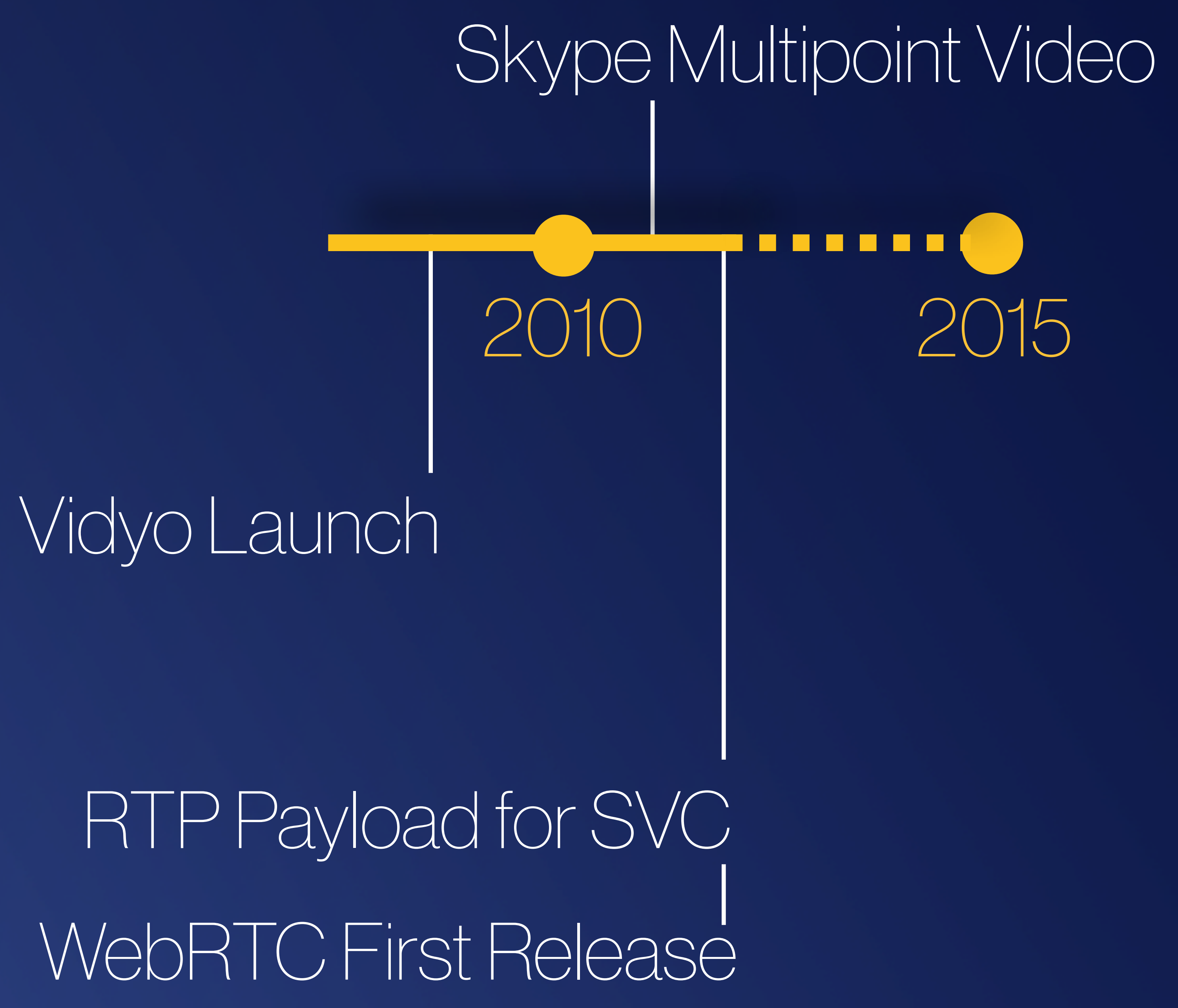
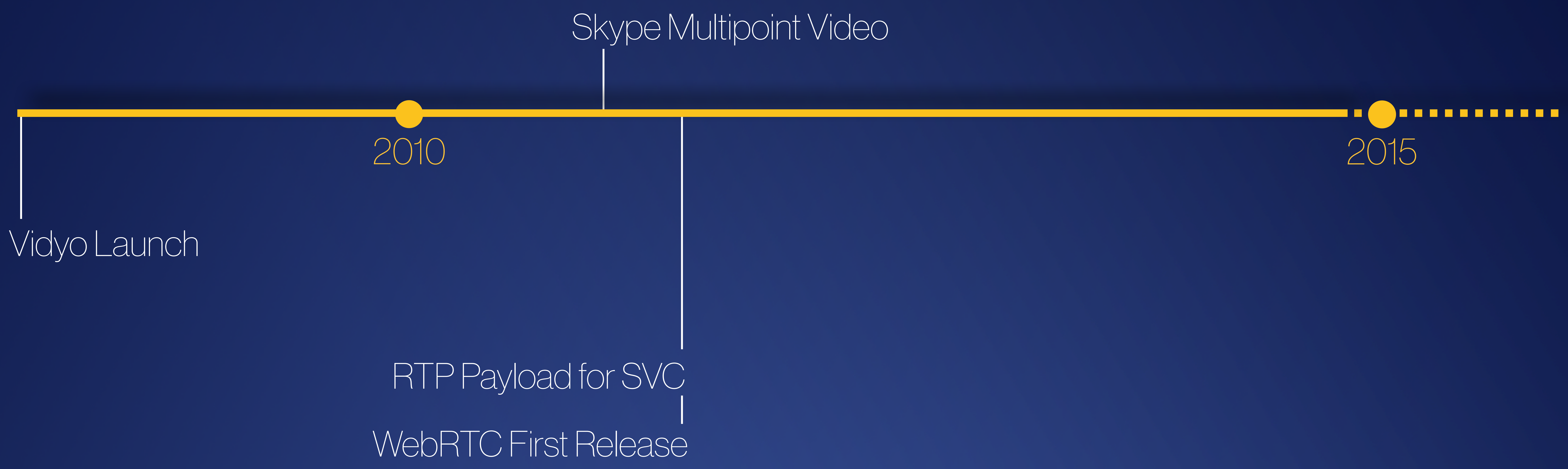
Chief Scientist & Co-founder


Skype Multipoint Video
2010
2015
Vidyo Launch
RTP Payload for SVC
WebRTC First Release



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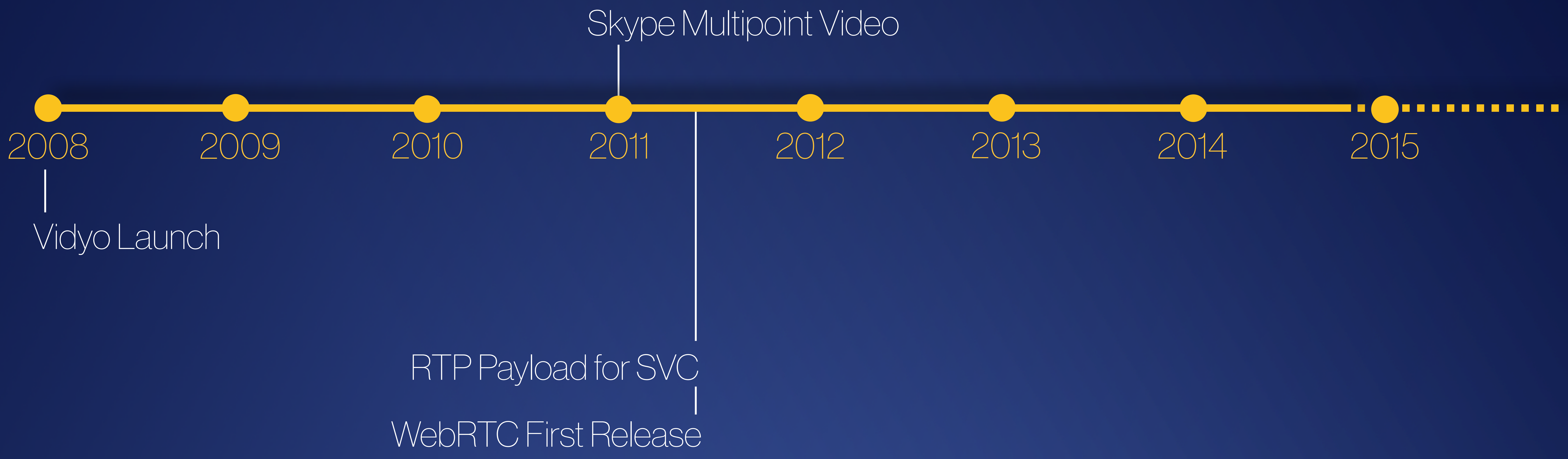
Chief Scientist & Co-founder



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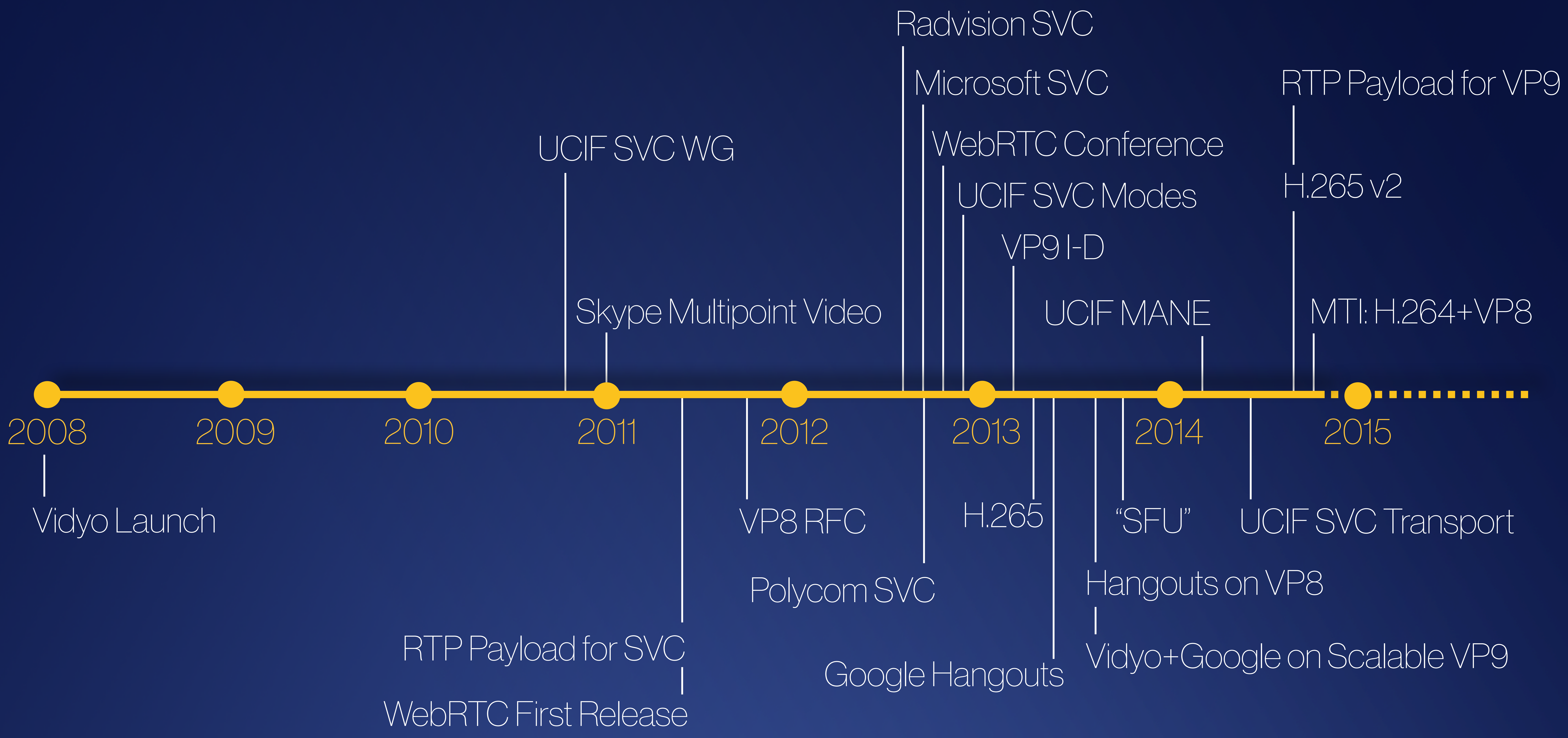
Chief Scientist & Co-founder



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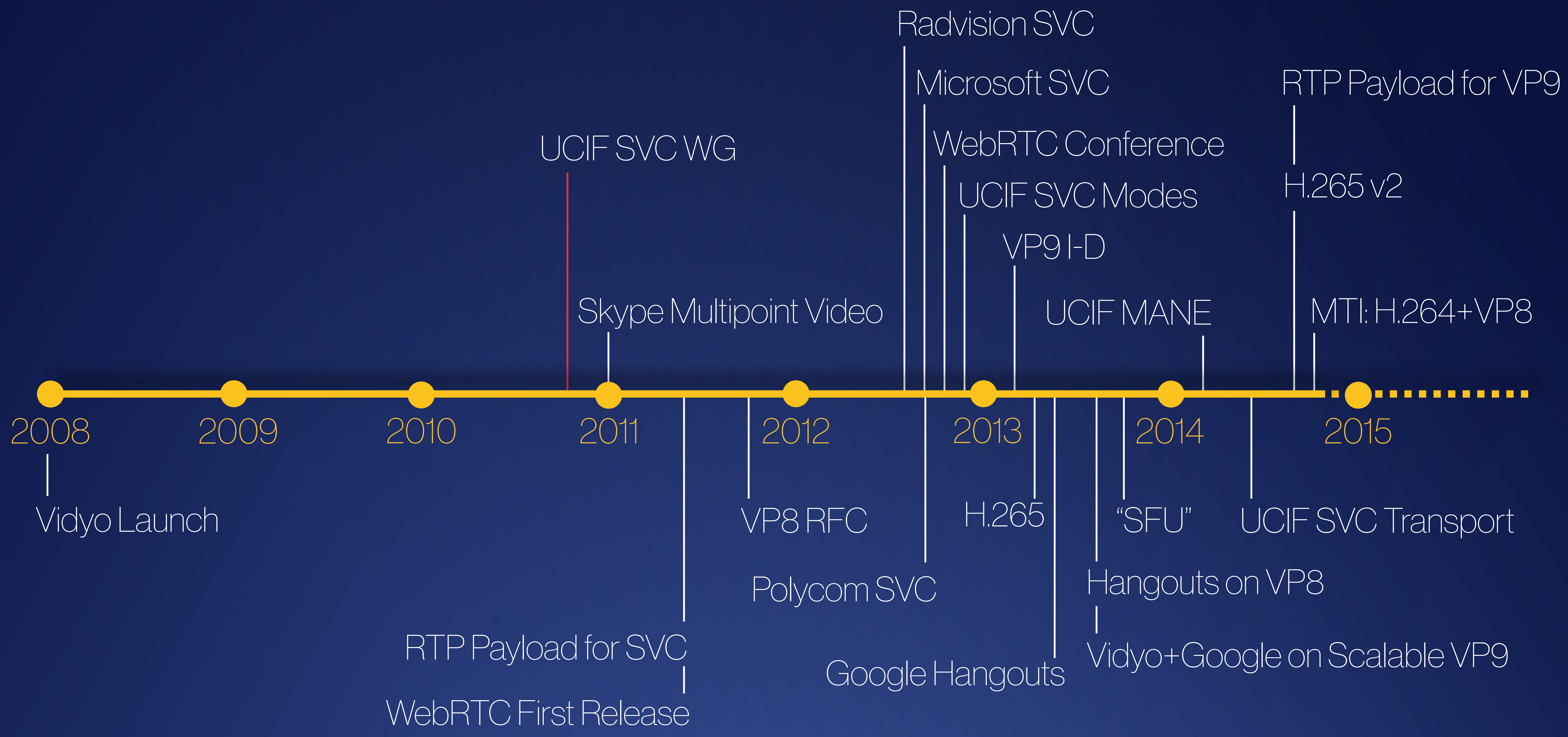
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



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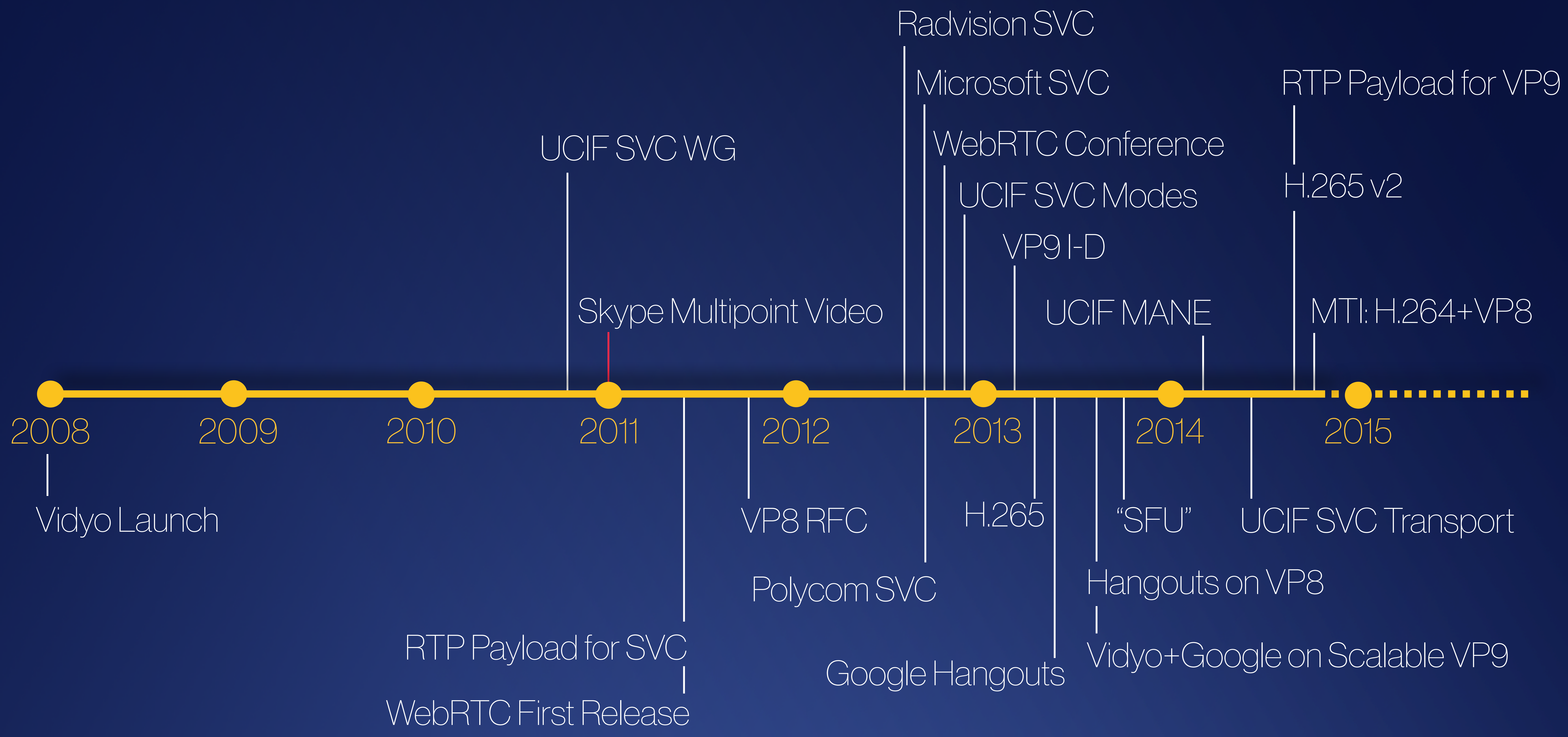
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

UCIF SVC WG

November 2010
UCI Forum SVC
Technical Working Group
formed

Certification and interop
for SVC systems



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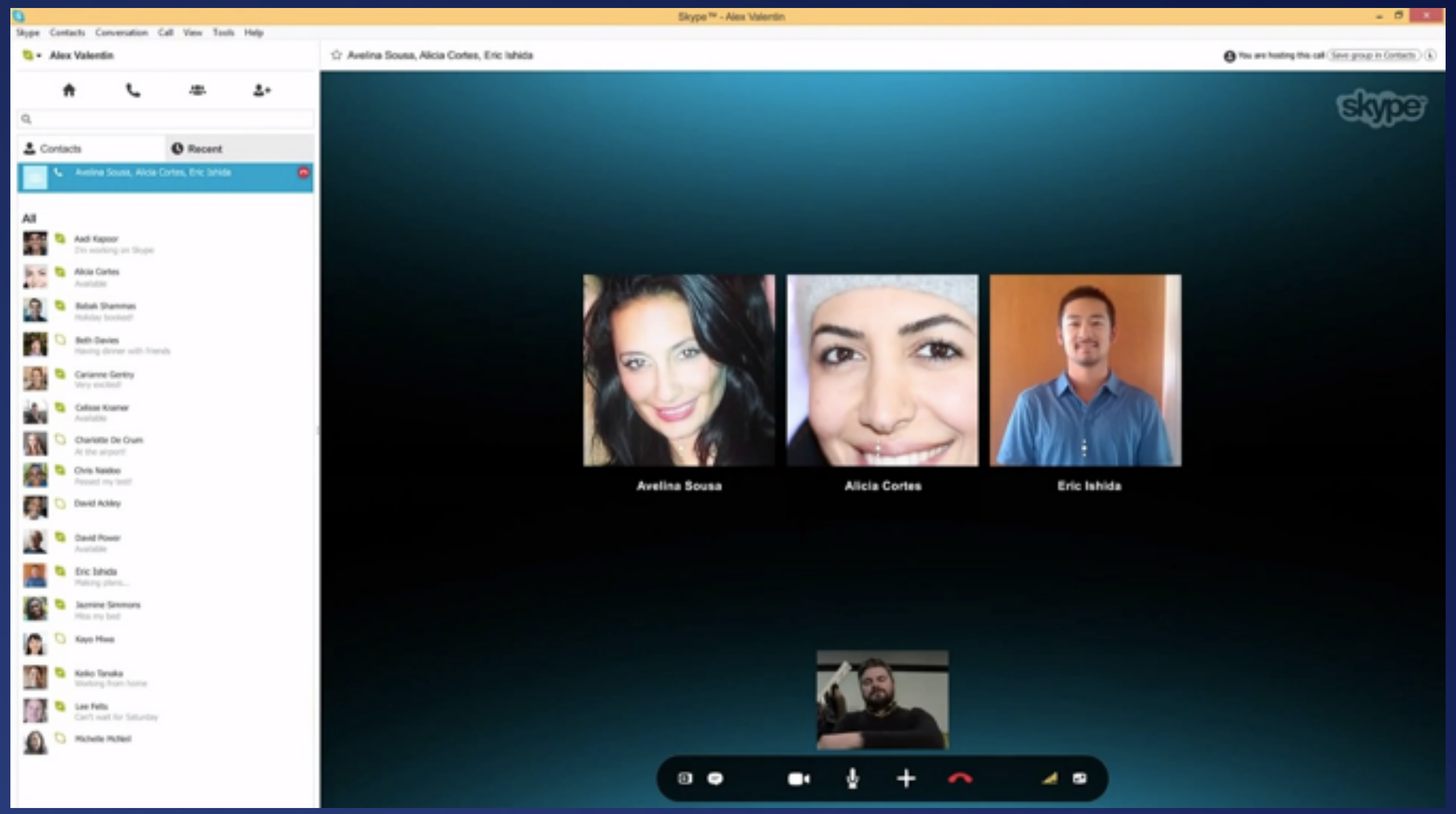
Chief Scientist & Co-founder

Skype Multipoint Video

January 2011

"Group Video Calling" with up to 5 participants as Premium service

Up to 10, and free as of April 2014



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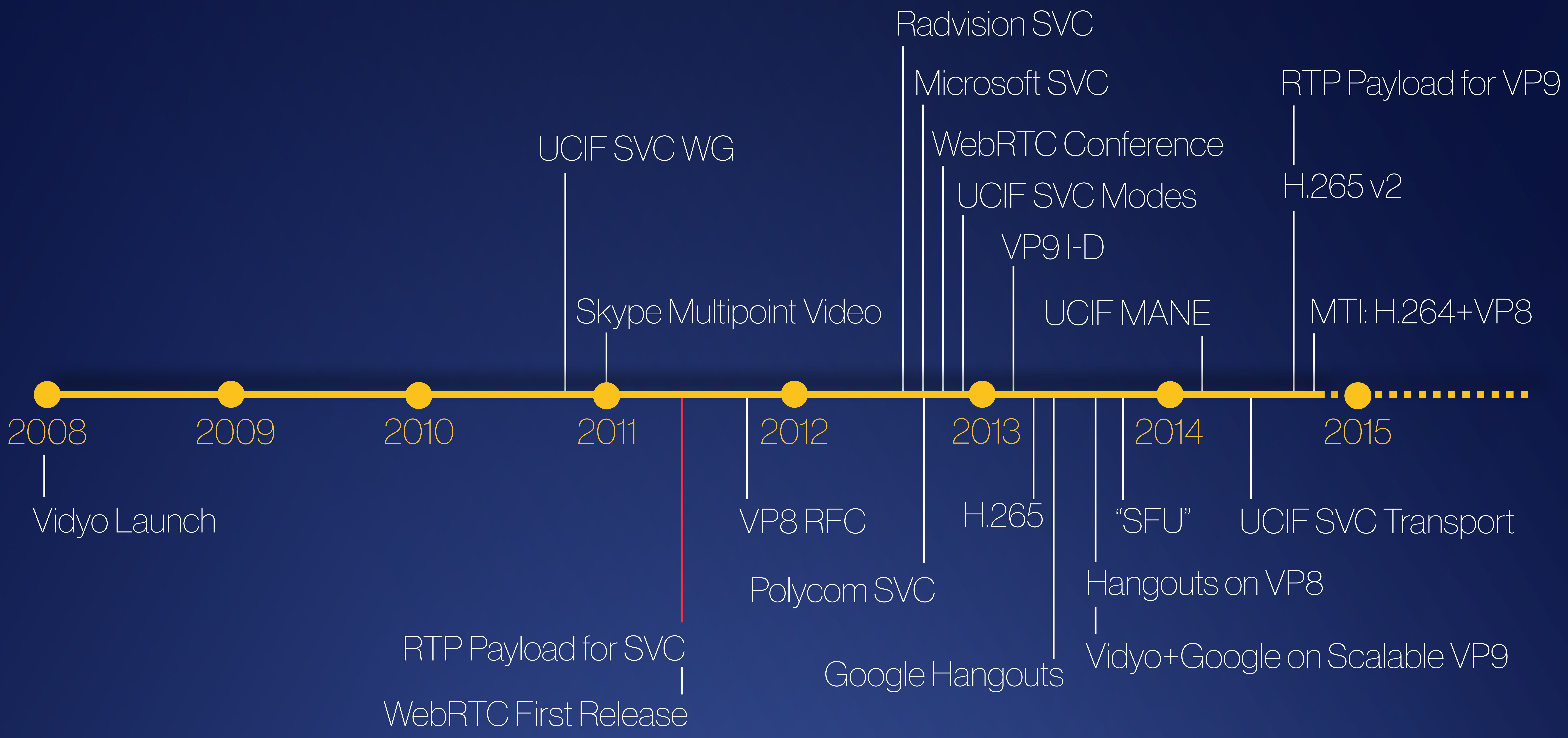
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

RTP Payload for SVC

May 2011
Wenger, Wang, Shierl,
and Eleftheriadis
RFC 6190

How to packetize H.264 SVC video
for RTP transport,
plus signaling parameters

```
Internet Engineering Task Force (IETF)                        S. Wenger
Request for Comments: 6190                                  Independent
Category: Standards Track                                     Y.-K. Wang
ISSN: 2070-1721                                     Huawei Technologies
                                                              T. Schierl
                                                           Fraunhofer HHI
                                                        A. Eleftheriadis
                                                                   Vidyo
                                                                May 2011


               RTP Payload Format for Scalable Video Coding

Abstract

   This memo describes an RTP payload format for Scalable Video Coding
   (SVC) as defined in Annex G of ITU-T Recommendation H.264, which is
   technically identical to Amendment 3 of ISO/IEC International
   Standard 14496-10.  The RTP payload format allows for packetization
   of one or more Network Abstraction Layer (NAL) units in each RTP
   packet payload, as well as fragmentation of a NAL unit in multiple
   RTP packets.  Furthermore, it supports transmission of an SVC stream
   over a single as well as multiple RTP sessions.  The payload format
   defines a new media subtype name "H264-SVC", but is still backward
   compatible to RFC 6184 since the base layer, when encapsulated in its
   own RTP stream, must use the H.264 media subtype name ("H264") and
   the packetization method specified in RFC 6184.  The payload format
   has wide applicability in videoconferencing, Internet video
   streaming, and high-bitrate entertainment-quality video, among
   others.

Status of This Memo

   This is an Internet Standards Track document.

   This document is a product of the Internet Engineering Task Force
   (IETF).  It represents the consensus of the IETF community.  It has
   received public review and has been approved for publication by the
   Internet Engineering Steering Group (IESG).  Further information on
   Internet Standards is available in Section 2 of RFC 5741.

   Information about the current status of this document, any errata,
   and how to provide feedback on it may be obtained at
   http://www.rfc-editor.org/info/rfc6190.



Wenger, et al.               Standards Track                    [Page 1]
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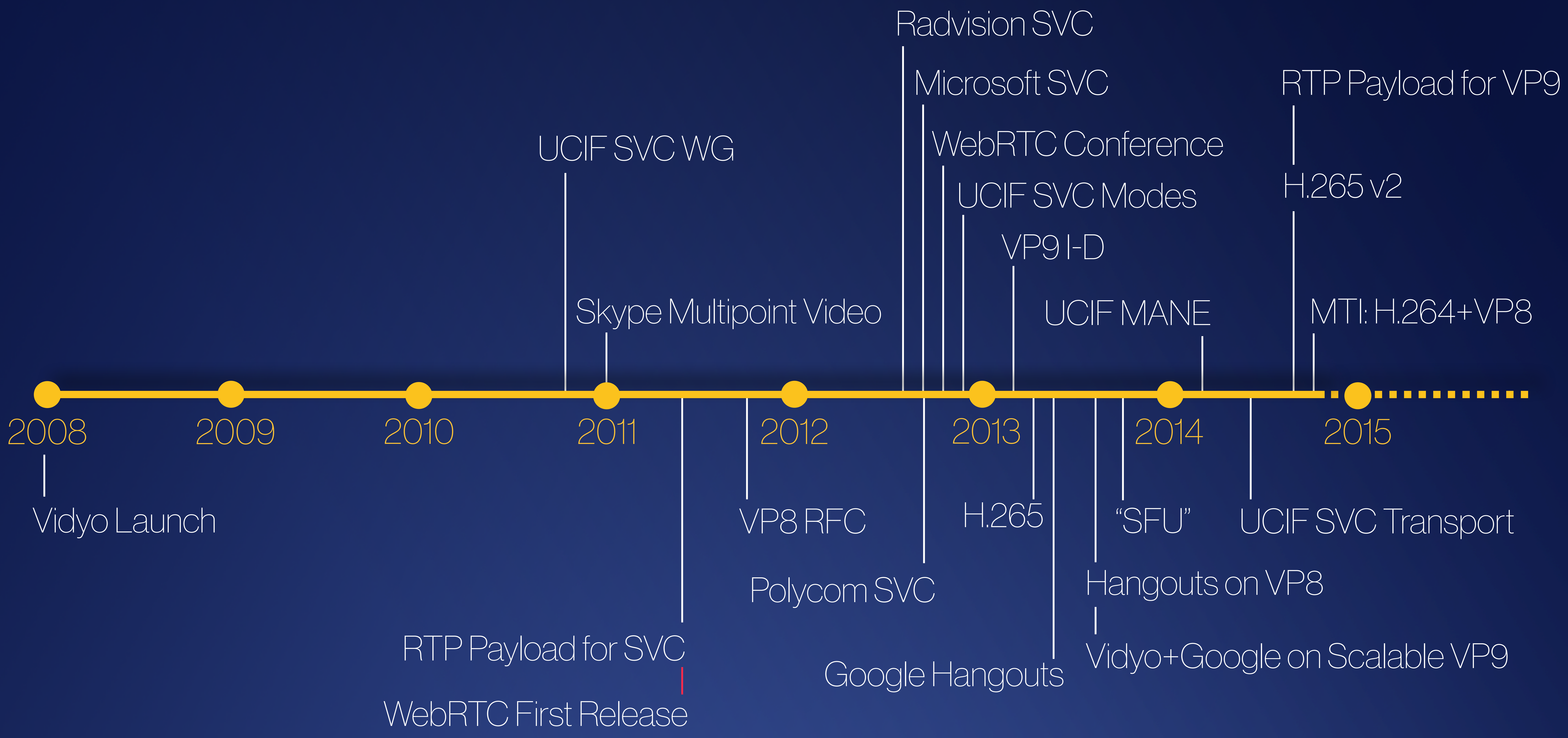


2008
Vidyo Launch
2009
2010
UCIF SVC WG
2011
Skype Multipoint Video
RTP Payload for SVC
WebRTC First Release
2012
VP8 RFC
Radvision SVC
Microsoft SVC
WebRTC Conference
UCIF SVC Modes
VP9 I-D
Polycom SVC
2013
H.265
Google Hangouts
UCIF MANE
Hangouts on VP8
Vidyo+Google on Scalable VP9
2014
"SFU"
UCIF SVC Transport
RTP Payload for VP9
H.265 v2
MTI: H.264+VP8
2015

WebRTC First Release

May 2011

Google releases WebRTC source code

Google release of WebRTC source code

From: Harald Alvestrand <hta@google.com>
Date: Wed, 1 Jun 2011 01:04:28 +0200
Message-ID: <BANLkTinXEJypAc2tVvAySzUGOtG9kLZNXA@mail.gmail.com>
To: public-webrtc@w3.org

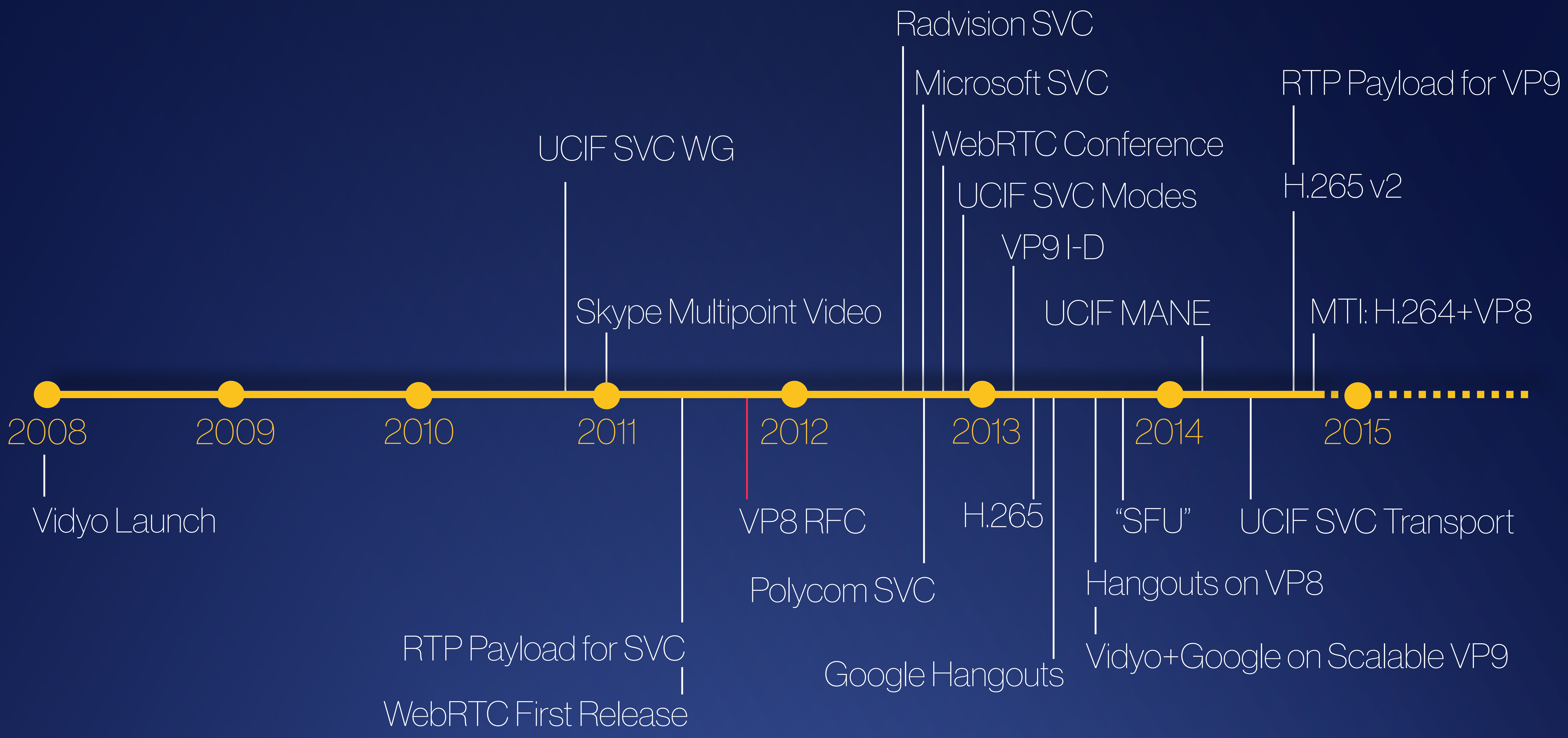
```
Today, Google made available WebRTC, an open source software package
for real-time voice and video on the web that we will be integrating
in Chrome.  This is our initial contribution to achieve the mission of
making audio and video available in all browsers, through a uniform
standard set of APIs.  This initial release will provide the
functionality we envision for WebRTC/RTCWEB, as detailed at
https://sites.google.com/site/webrtc/.  Working with the browser
community and working groups like this, our goal is to expand the
available APIs over the next few months for developers to create voice
and video applications on the web.

The underlying components we're releasing are stable and the
interfaces for this initial release are consistent with the
discussions in this working group. We will continue to provide working
implementations for consideration and feedback to collectively ensure
stable standards are finalized.  Google is committed to fully
supporting these standards and we look forward to your input in the
coming months.


                   Harald, speaking for Google.
```

Received on Tuesday, 31 May 2011 23:05:13 GMT

This archive was generated by hypermail 2.2.0+W3C-0.50 : Tuesday, 31 May 2011 23:05:14 GMT

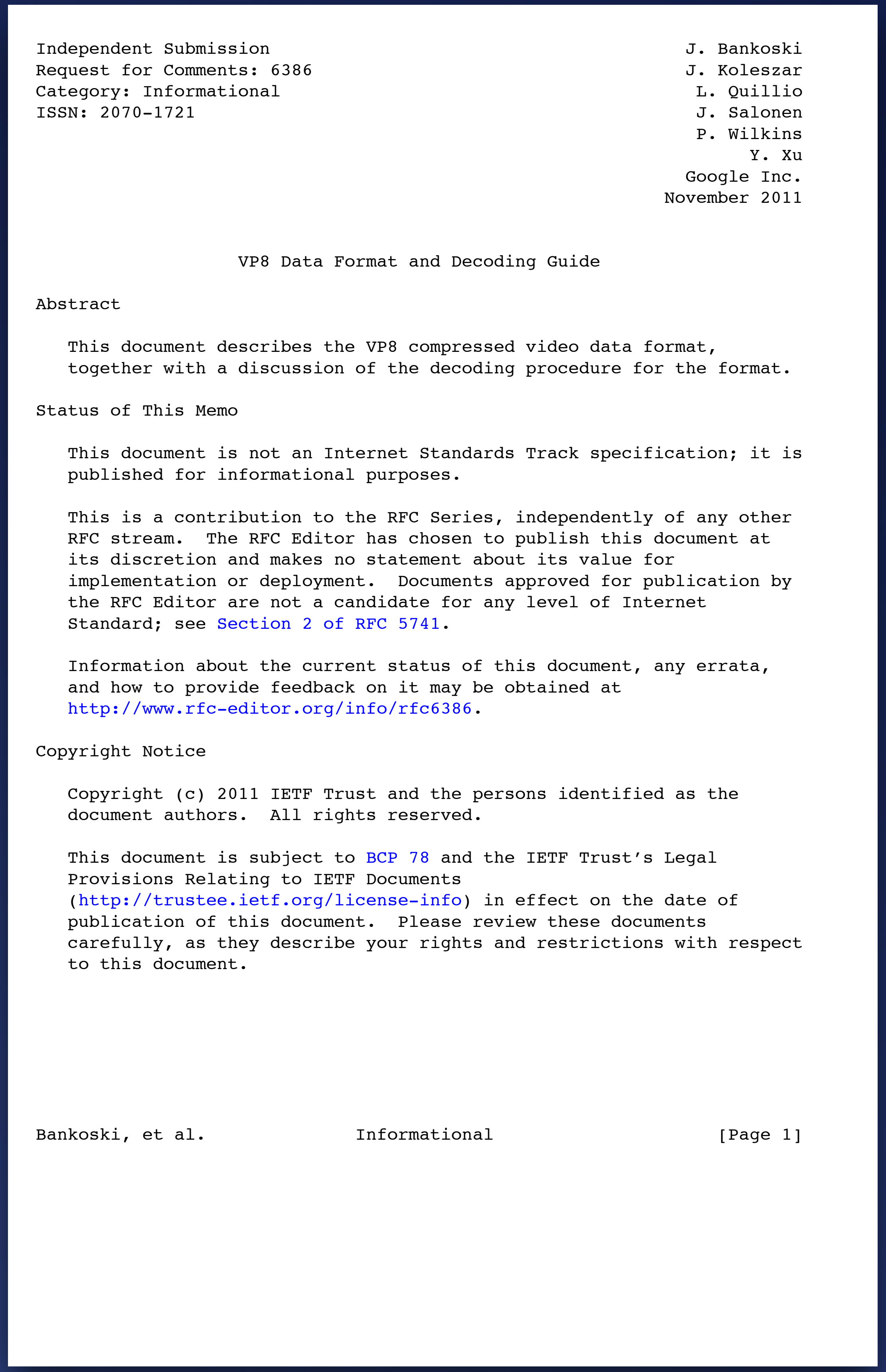


Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

VP8 RFC

November 2011
Bankoski, Koleszar, Quillio, Salonen, Wilkins, and Xu
RFC 6386

VP8 data format and decoding guide (independent submission)



Independent Submission — J. Bankoski
Request for Comments: 6386 — J. Koleszar
Category: Informational — L. Quillio
ISSN: 2070-1721 — J. Salonen
P. Wilkins
Y. Xu
Google Inc.
November 2011

VP8 Data Format and Decoding Guide

Abstract

This document describes the VP8 compressed video data format, together with a discussion of the decoding procedure for the format.

Status of This Memo

This document is not an Internet Standards Track specification; it is published for informational purposes.

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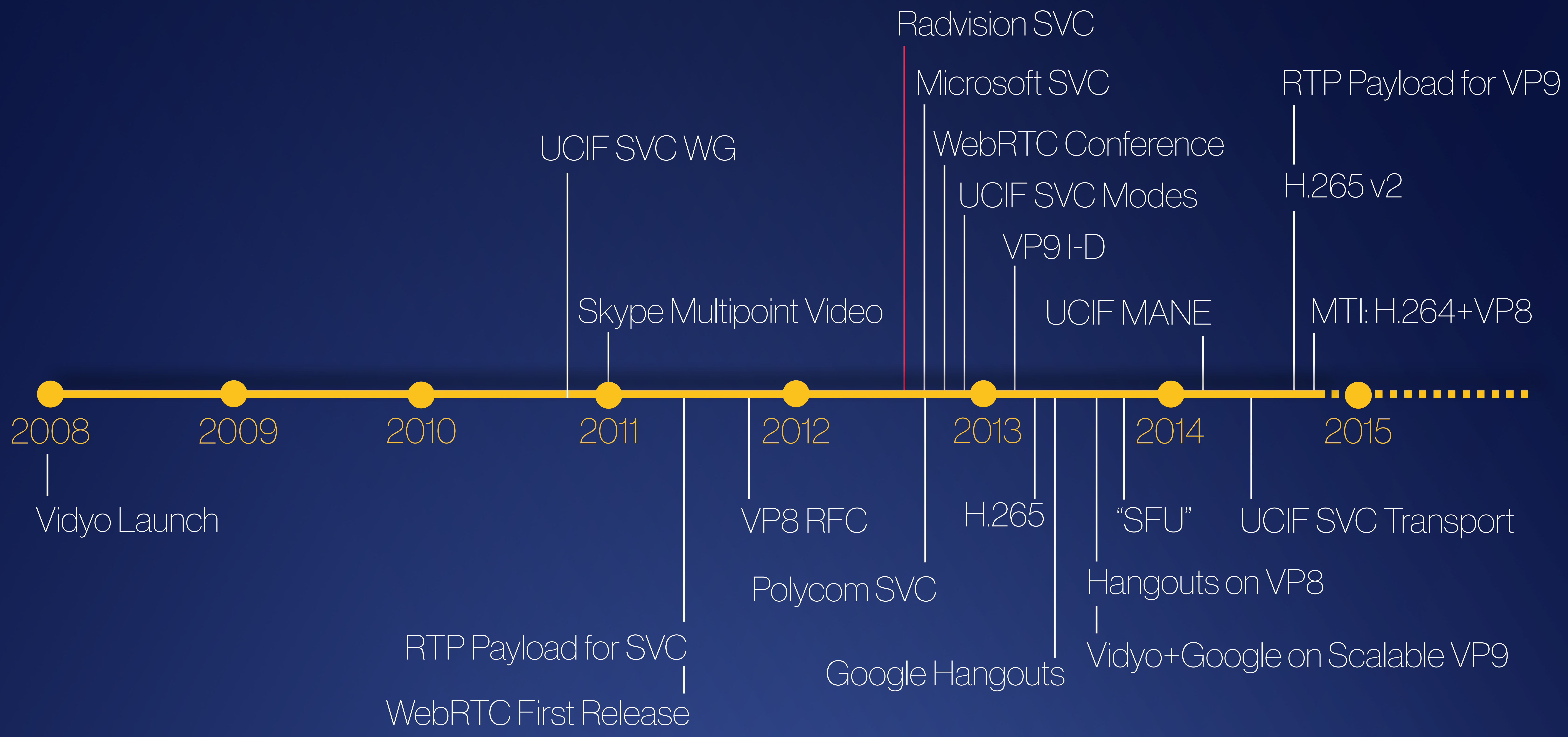
Bankoski, et al. — Informational — [Page 1]

Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Radvision SVC

September 2012

Elite 5000 Series MCU Ver. 7.7
SVC for error resilience


RADVISION®
Delivering the Visual Experience®

SCOPIA®
INFRASTRUCTURE

SCOPIA Elite MCU 5000 Series

SCOPIA Elite Provides the Highest Quality Experience and Most Advanced Conferencing Capabilities Available

Supreme High Definition
With 1080p High Definition processing, multistream telepresence connectivity, unlimited conferences, personal video layout per participant and AAC audio, SCOPIA Elite delivers supreme High Definition conferencing. Utilization of the very latest state of the art DSP technology offers uncompromised HD.

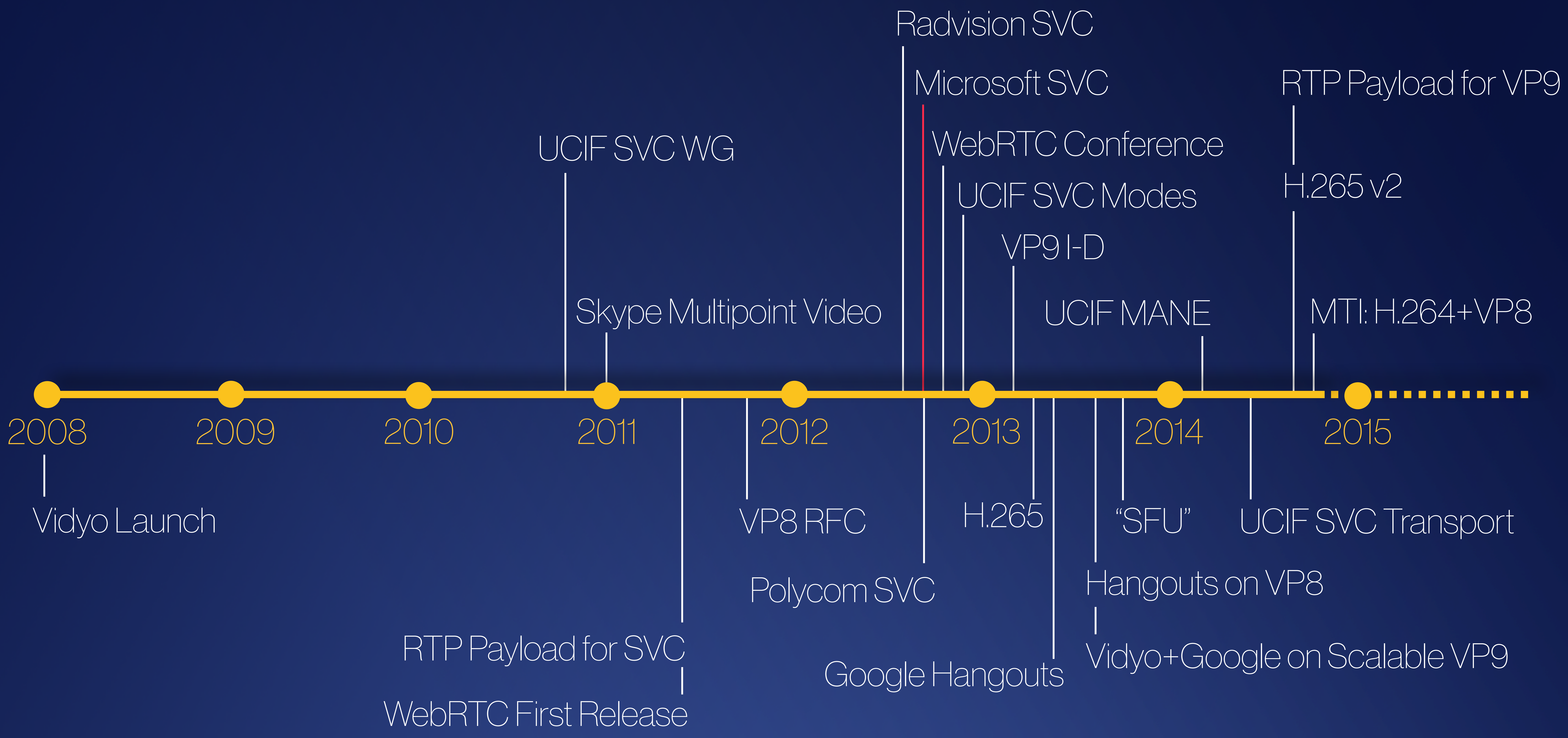
Flexible Resources
Mixed endpoint applications are supported with ease and efficiency. Enhanced Definition at 4X capacity affords excellent value. On-demand, dynamic and automatic resource allocation supports any combination of endpoint capabilities without complex configuration changes.

Optimized Quality
Encoder per participant ensures an optimal quality experience for any endpoint on any network. H.264 Scalable Video Coding (SVC) delivers superior performance over compromised networks.

Maximum Usability
SCOPIA Elite was designed by users and administrators for usability and simplicity. On-screen information overlays, easy conference creation and entry through the enhanced video auto attendant and IVR make attending conferences simple. The industry leading management interface makes seemingly complicated tasks simple; users can view important statistics at a glance saving time, money and headache.

Designed to Scale
Unequalled scalability is achieved through patented, distributed multipoint conferencing. SCOPIA Elite utilizes SVC, enabling enhanced connections between MCUs in poor network conditions. The Advanced Telecommunications Computing Architecture (ATCA) delivers investment protection through a future proof platform.

Vidyo®
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



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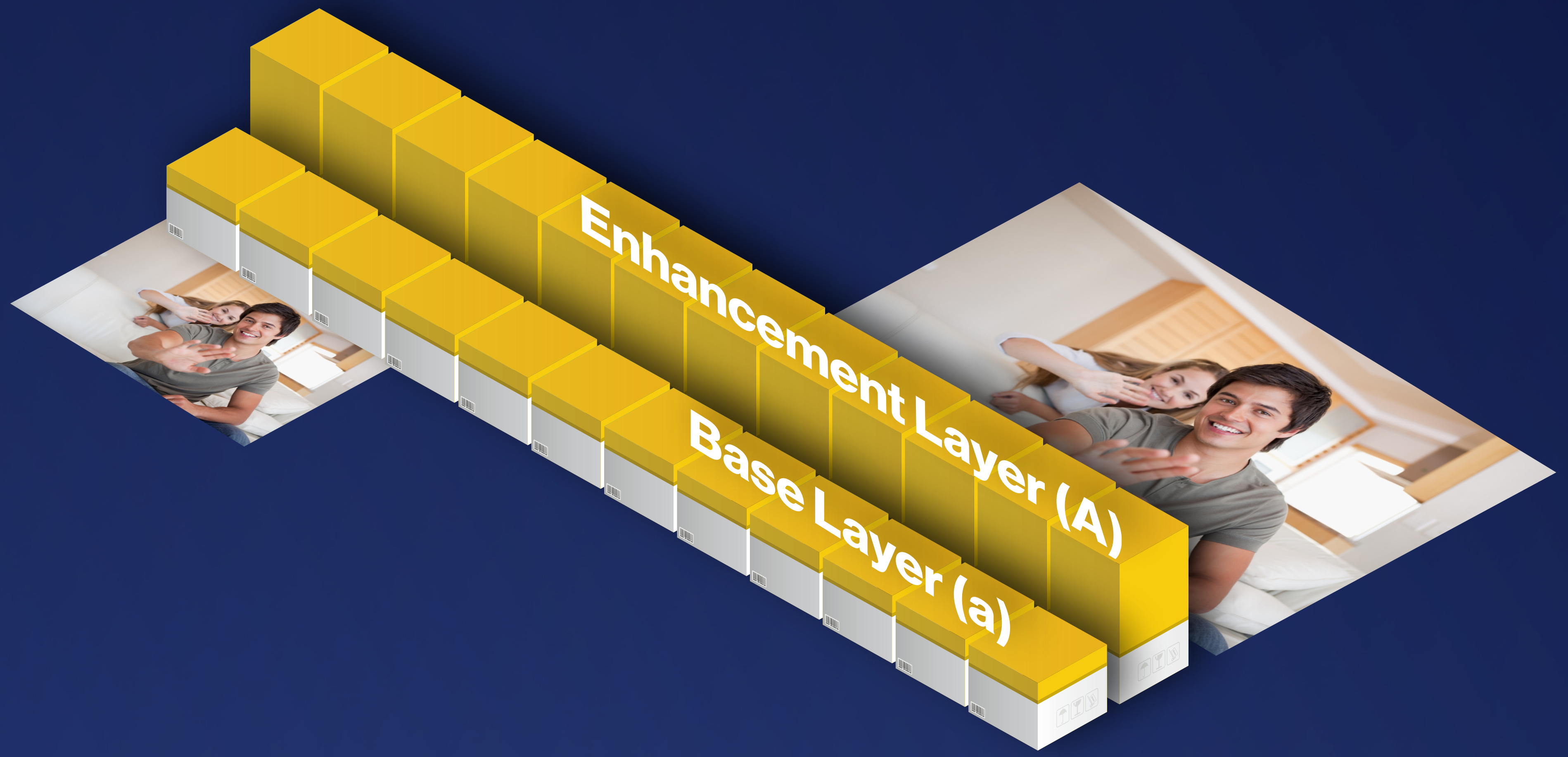
Chief Scientist & Co-founder

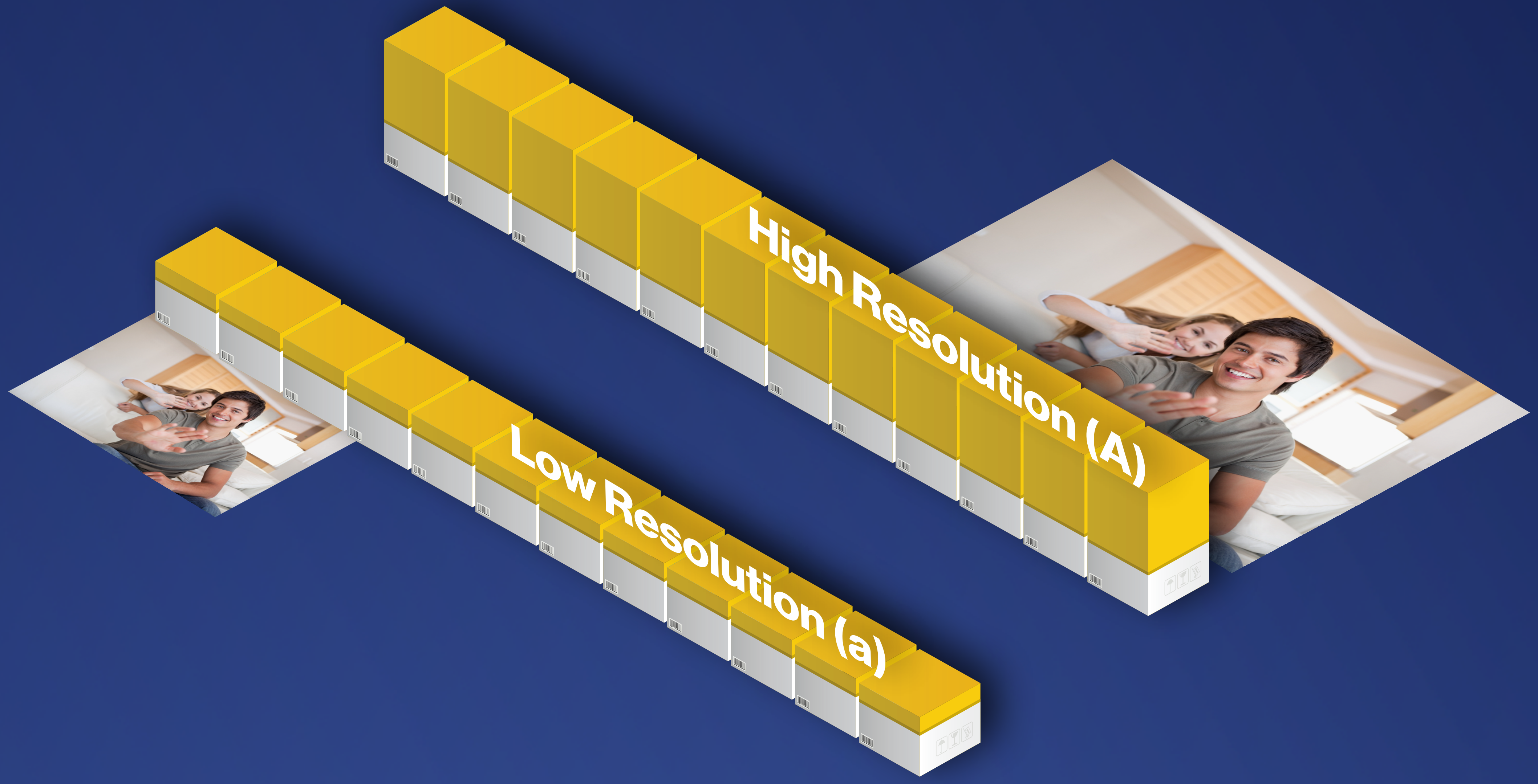
Microsoft SVC

October 2012

Simulcast SVC (temporal only)
on Lync 2013

Simulcasting vs. Scalable Coding





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Chief Scientist & Co-founder

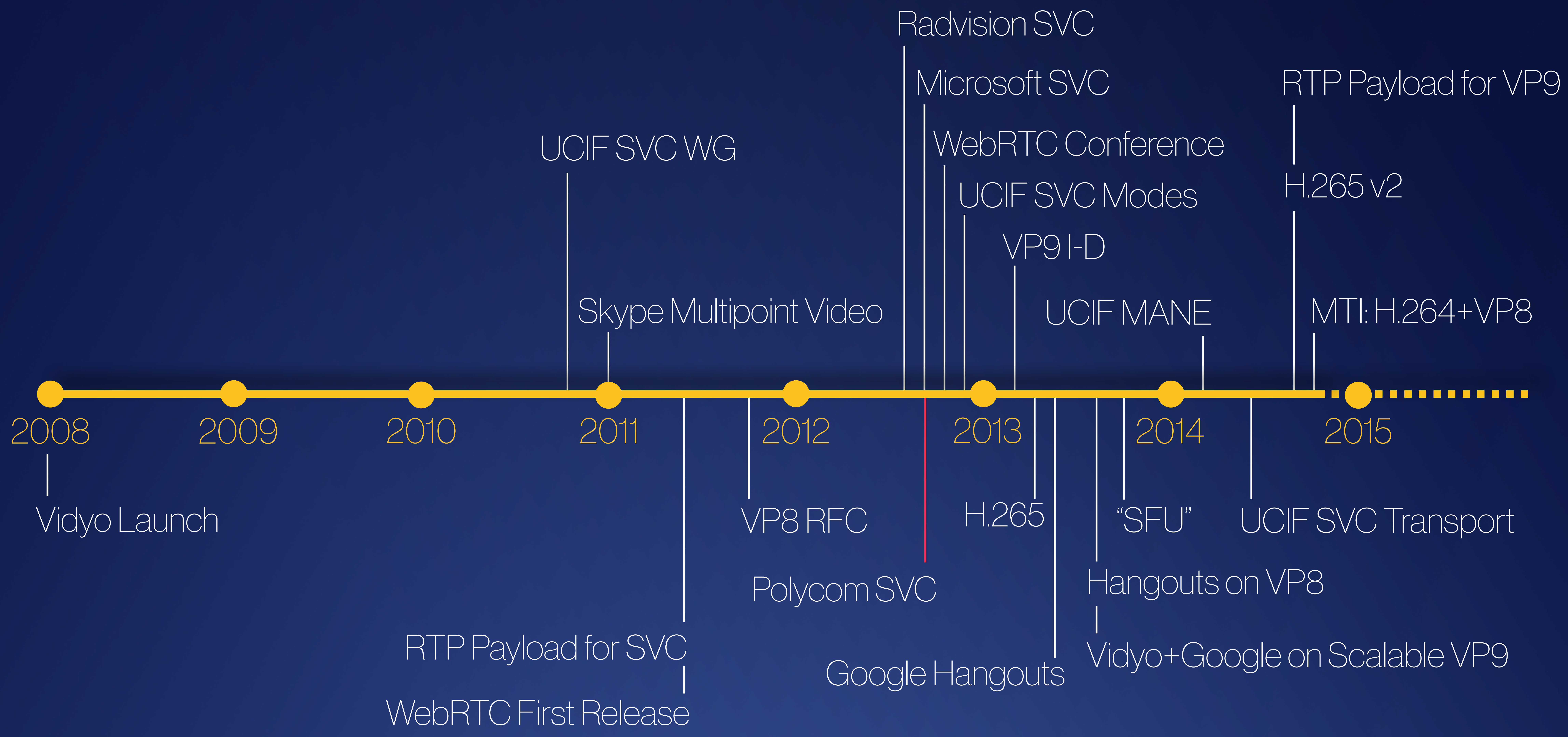
Simulcast Server

Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

"Selective Forwarding Unit"

SFU



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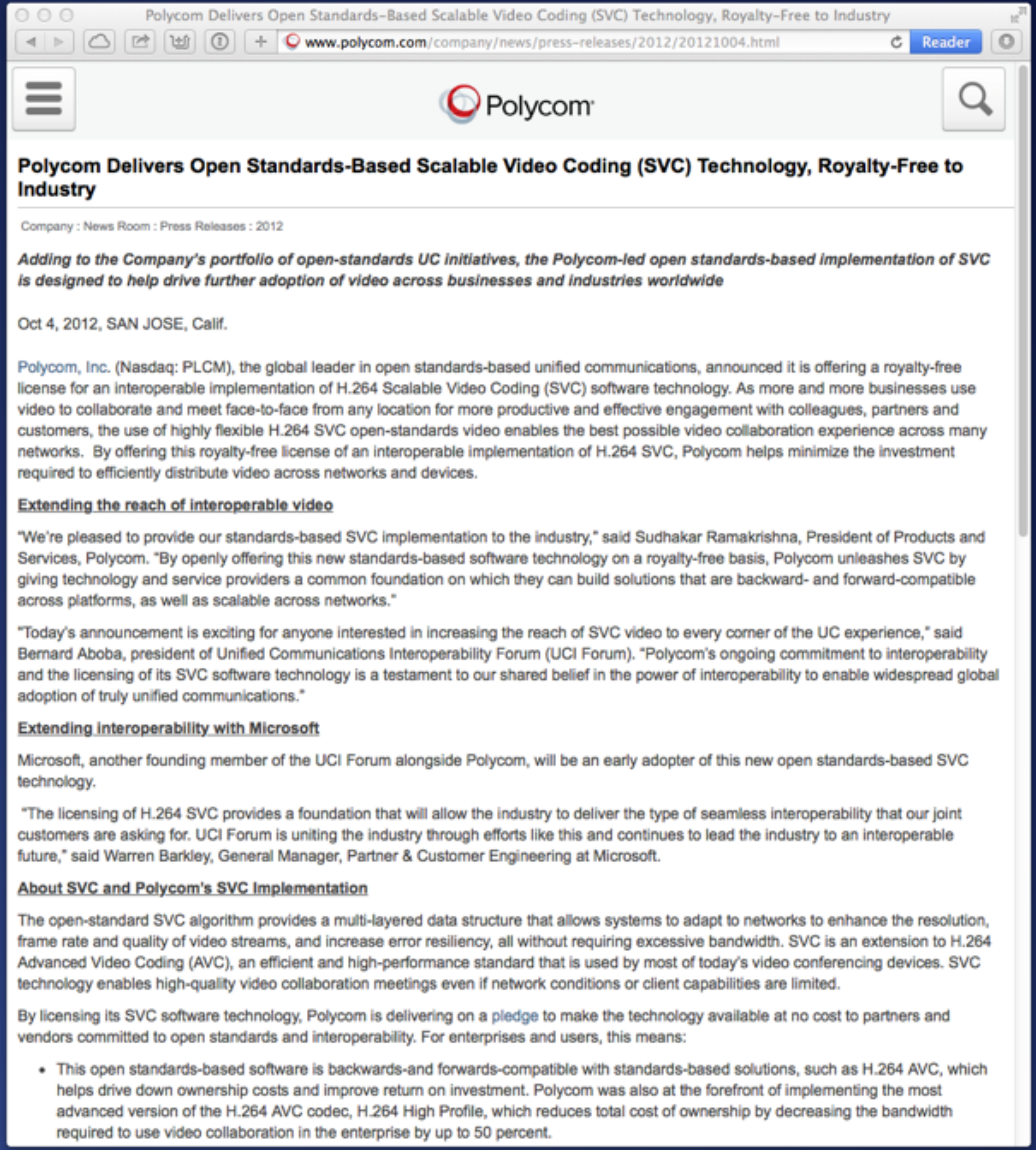
Chief Scientist & Co-founder

Polycom SVC

October 2012

Announces availability of royalty-free SVC implementation, with Microsoft as early adopter



Polycom Delivers Open Standards-Based Scalable Video Coding (SVC) Technology, Royalty-Free to Industry

www.polycom.com/company/news/press-releases/2012/20121004.html

Polycom

Polycom Delivers Open Standards-Based Scalable Video Coding (SVC) Technology, Royalty-Free to Industry

Company : News Room : Press Releases : 2012

Adding to the Company's portfolio of open-standards UC initiatives, the Polycom-led open standards-based implementation of SVC is designed to help drive further adoption of video across businesses and industries worldwide

Oct 4, 2012, SAN JOSE, Calif.

Polycom, Inc. (Nasdaq: PLCM), the global leader in open standards-based unified communications, announced it is offering a royalty-free license for an interoperable implementation of H.264 Scalable Video Coding (SVC) software technology. As more and more businesses use video to collaborate and meet face-to-face from any location for more productive and effective engagement with colleagues, partners and customers, the use of highly flexible H.264 SVC open-standards video enables the best possible video collaboration experience across many networks. By offering this royalty-free license of an interoperable implementation of H.264 SVC, Polycom helps minimize the investment required to efficiently distribute video across networks and devices.

Extending the reach of interoperable video

"We're pleased to provide our standards-based SVC implementation to the industry," said Sudhakar Ramakrishna, President of Products and Services, Polycom. "By openly offering this new standards-based software technology on a royalty-free basis, Polycom unleashes SVC by giving technology and service providers a common foundation on which they can build solutions that are backward- and forward-compatible across platforms, as well as scalable across networks."

"Today's announcement is exciting for anyone interested in increasing the reach of SVC video to every corner of the UC experience," said Bernard Aboba, president of Unified Communications Interoperability Forum (UCI Forum). "Polycom's ongoing commitment to interoperability and the licensing of its SVC software technology is a testament to our shared belief in the power of interoperability to enable widespread global adoption of truly unified communications."

Extending interoperability with Microsoft

Microsoft, another founding member of the UCI Forum alongside Polycom, will be an early adopter of this new open standards-based SVC technology.

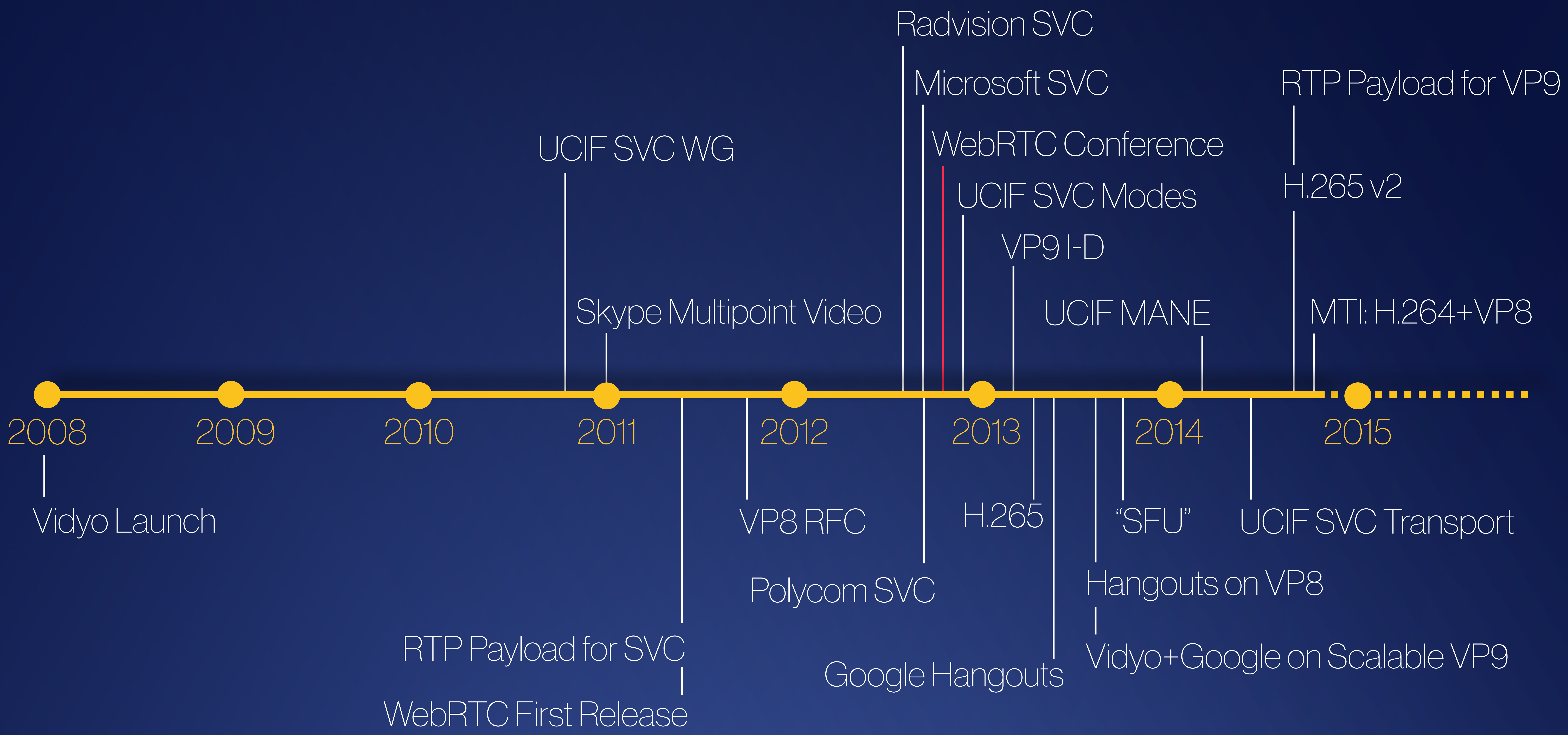
"The licensing of H.264 SVC provides a foundation that will allow the industry to deliver the type of seamless interoperability that our joint customers are asking for. UCI Forum is uniting the industry through efforts like this and continues to lead the industry to an interoperable future," said Warren Barkley, General Manager, Partner & Customer Engineering at Microsoft.

About SVC and Polycom's SVC Implementation

The open-standard SVC algorithm provides a multi-layered data structure that allows systems to adapt to networks to enhance the resolution, frame rate and quality of video streams, and increase error resiliency, all without requiring excessive bandwidth. SVC is an extension to H.264 Advanced Video Coding (AVC), an efficient and high-performance standard that is used by most of today's video conferencing devices. SVC technology enables high-quality video collaboration meetings even if network conditions or client capabilities are limited.

By licensing its SVC software technology, Polycom is delivering on a pledge to make the technology available at no cost to partners and vendors committed to open standards and interoperability. For enterprises and users, this means:

- This open standards-based software is backwards-and forwards-compatible with standards-based solutions, such as H.264 AVC, which helps drive down ownership costs and improve return on investment. Polycom was also at the forefront of implementing the most advanced version of the H.264 AVC codec, H.264 High Profile, which reduces total cost of ownership by decreasing the bandwidth required to use video collaboration in the enterprise by up to 50 percent.



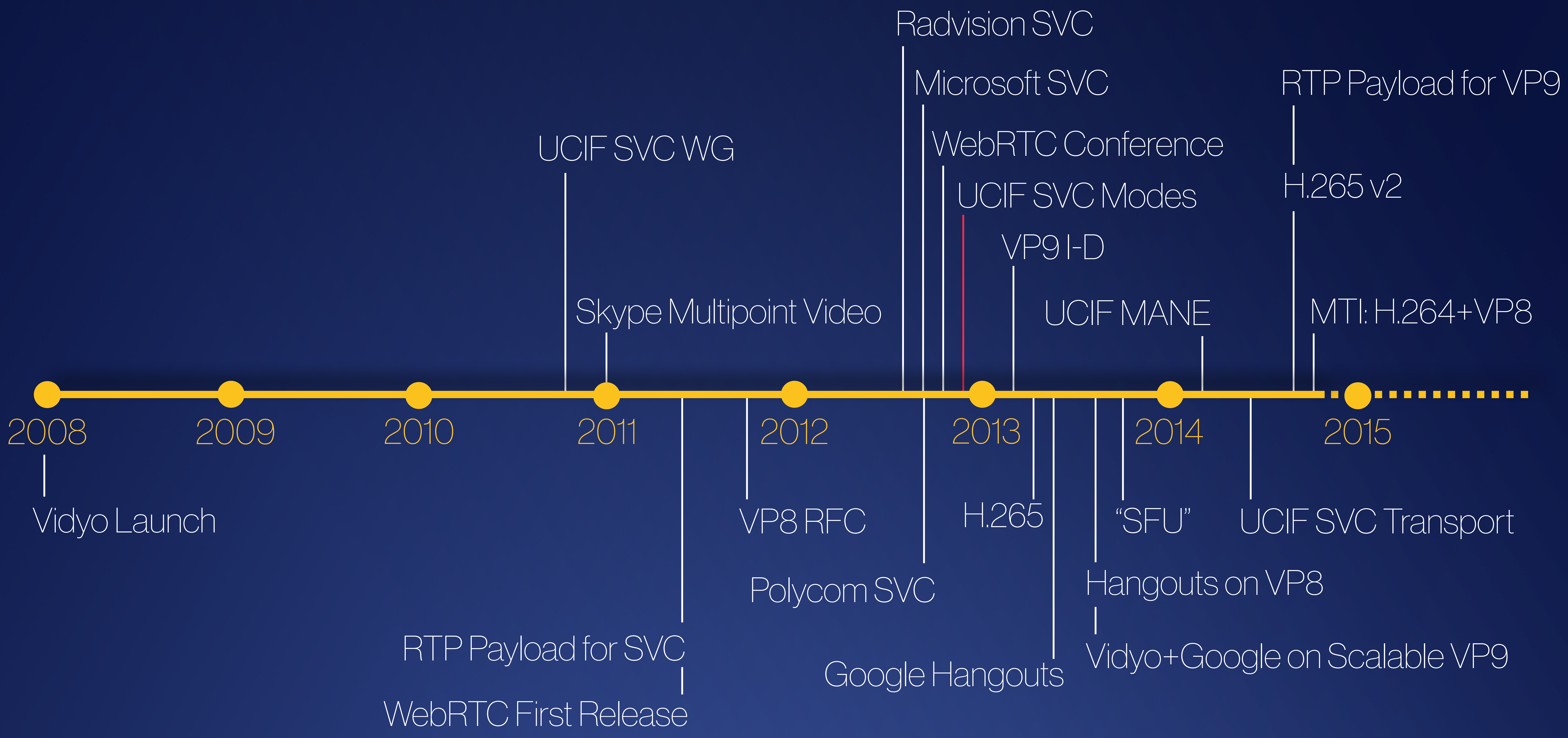
Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

WebRTC Conference

November 27-29, 2012

First WebRTC Conference and Expo
South San Francisco

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



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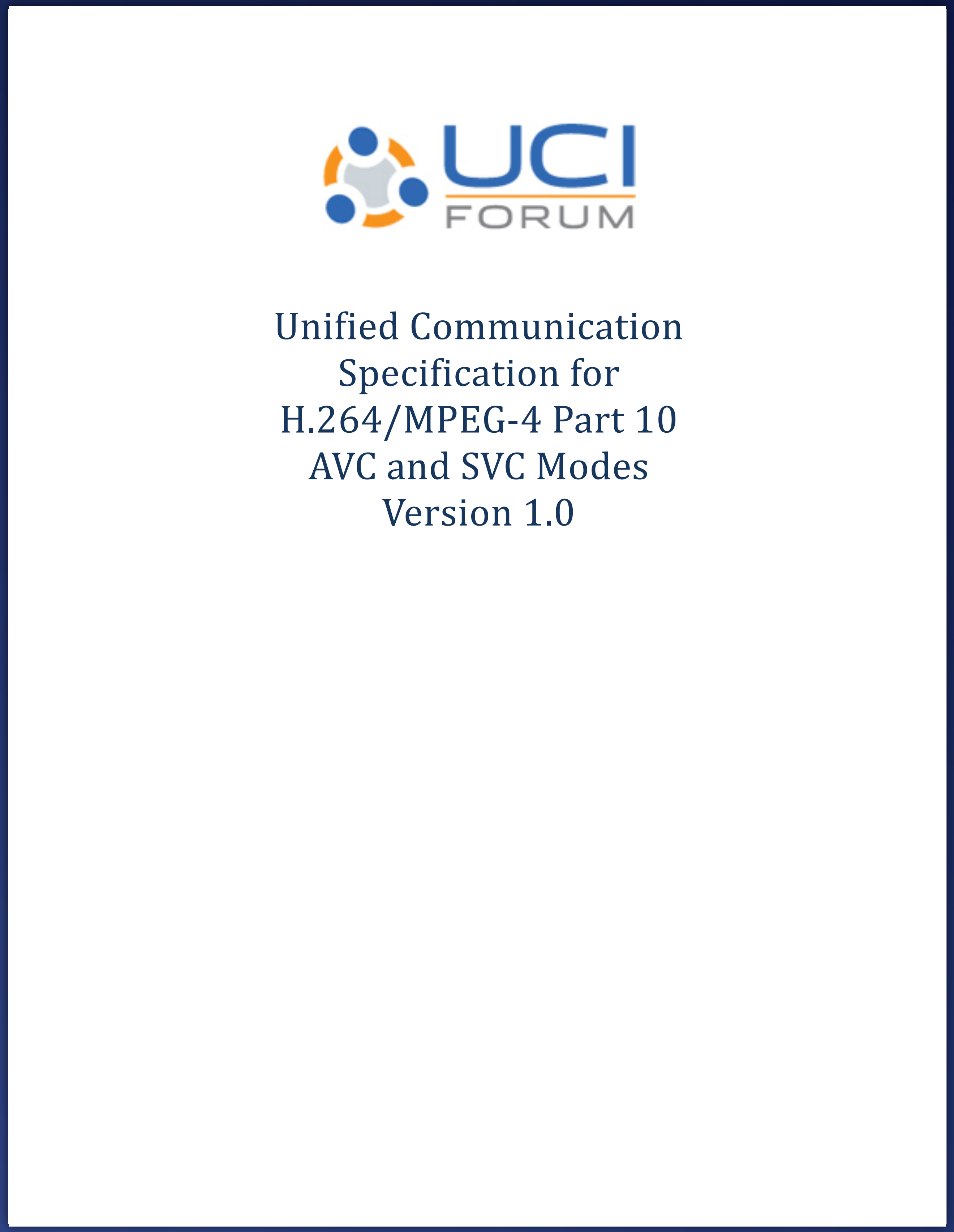
Chief Scientist & Co-founder

UCIF SVC Modes

December 2012

AVC and SVC Video
Bitstream Modes

(Lync does Mode 1, Vidyo Mode 2s)


UCI FORUM

Unified Communication
Specification for
H.264/MPEG-4 Part 10
AVC and SVC Modes
Version 1.0

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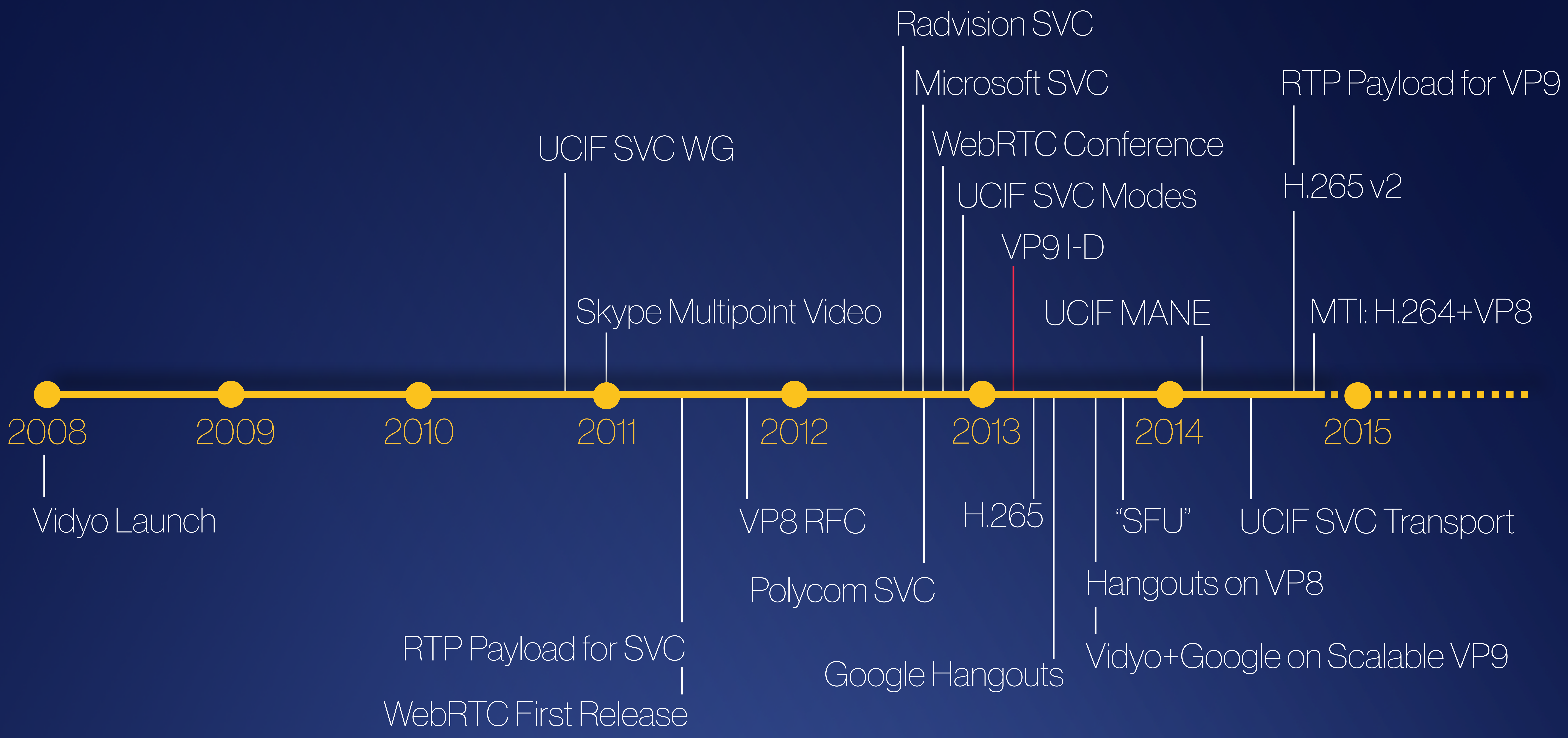
Chief Scientist & Co-founder



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Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

VP9 I-D

February 2013
Grange and Alvestrand
draft-grange-vp9-bitstream

Overview of VP9 Bitstream

```
Network Working Group                                          A. Grange
Internet-Draft                                             H. Alvestrand
Intended status: Informational                                    Google
Expires: August 22, 2013                               February 18, 2013


                       A VP9 Bitstream Overview
                    draft-grange-vp9-bitstream-00

Abstract

   This document describes VP9, a video codec being developed
   specifically to meet the demand for the consumption of video over the
   Internet, including professionally and amateur produced video-on-
   demand and conversational video content.  VP9 is an evolution of the
   VP8 video codec that is described in [bankoski-rfc6386] and includes
   a number of enhancements and new coding tools that have been added to
   improve the coding efficiency.  The new tools that have been added so
   far include: larger prediction block sizes up to 64x64, various forms
   of compound INTER prediction, more modes for INTRA prediction,
   &#8539;-pel motion vectors, 8-tap switchable sub-pixel interpolation
   filters, improved motion reference generation, improved motion vector
   coding, improved entropy coding including frame-level entropy
   adaptation for various symbols, improved loop filtering, the
   incorporation of the Asymmetric Discrete Sine Transform (ADST),
   larger 16x16 and 32x32 DCTs, and improved frame level segmentation.
   VP9 is under active development and this document provides only a
   snapshot of the current state of the coding tools as they exist
   today.  The finalized version of the VP9 bitstream may differ
   considerably from the description contained herein and may encompass
   the exclusion or modification of existing coding tools or the
   addition of new coding tools.

Status of this Memo

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Grange & Alvestrand      Expires August 22, 2013                [Page 1]
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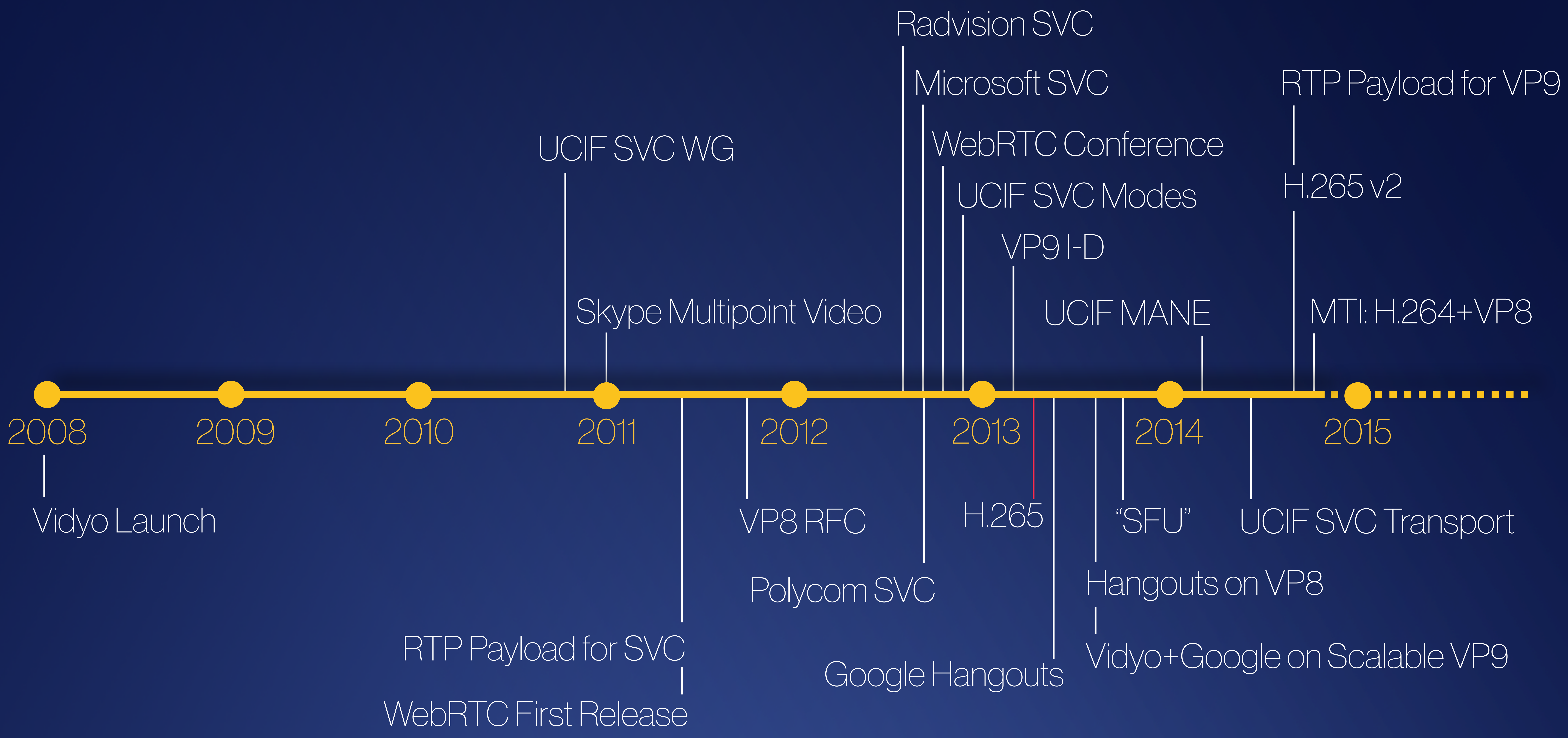
Chief Scientist & Co-founder

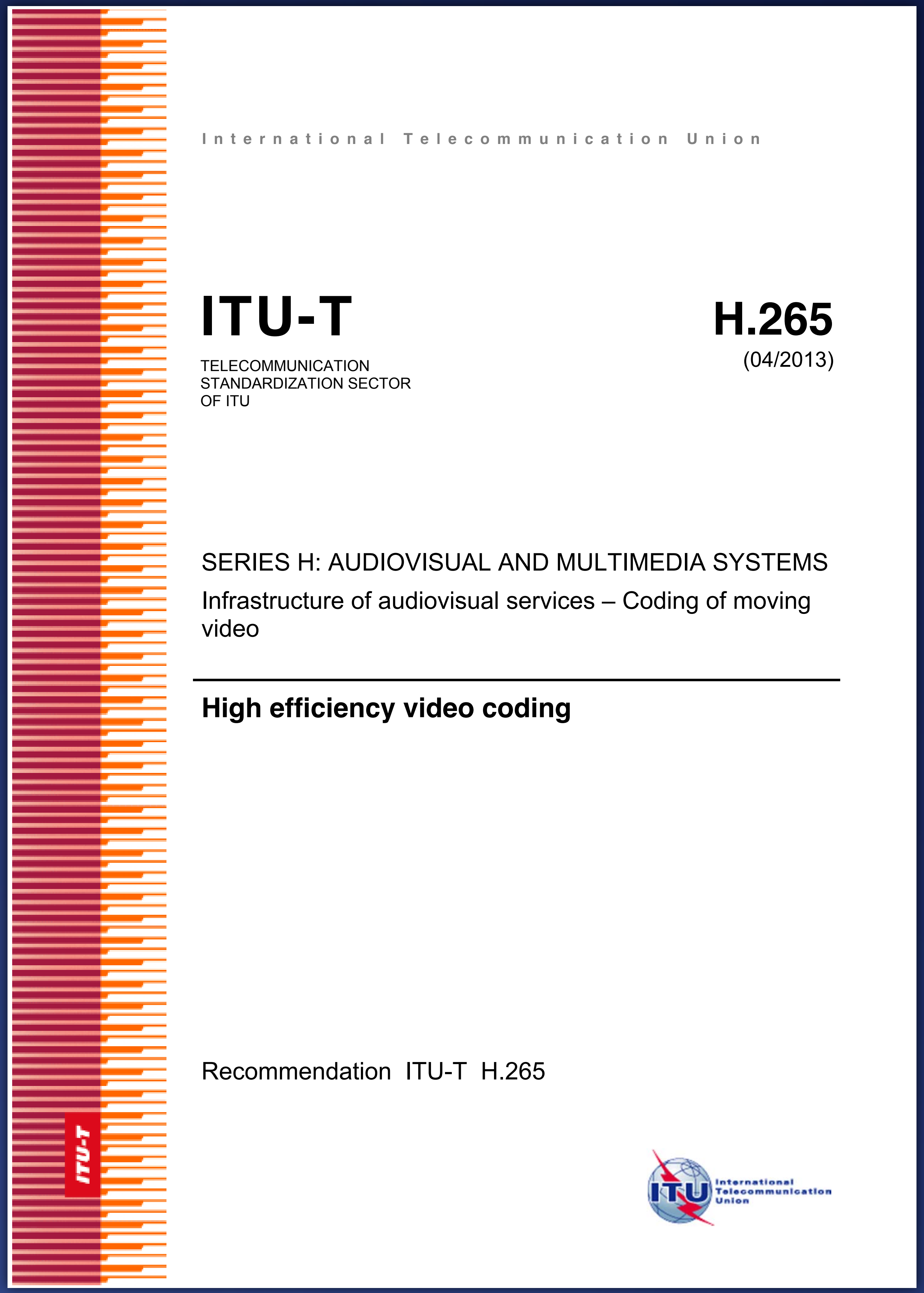

Radvision SVC
Microsoft SVC
RTP Payload for VP9
WebRTC Conference
UCIF SVC WG
UCIF SVC Modes
H.265 v2
VP9 I-D
Skype Multipoint Video
UCIF MANE
MTI: H.264+VP8
2008
2009
2010
2011
2012
2013
2014
2015
Vidyo Launch
VP8 RFC
H.265
"SFU"
UCIF SVC Transport
Polycom SVC
Hangouts on VP8
RTP Payload for SVC
Google Hangouts
Vidyo+Google on Scalable VP9
WebRTC First Release

H.265

April 2013

HEVC - High Efficiency Video Coding

50% bitrate reduction from H.264
ideal for 1080p and 4K
built-in temporal scalability


International Telecommunication Union

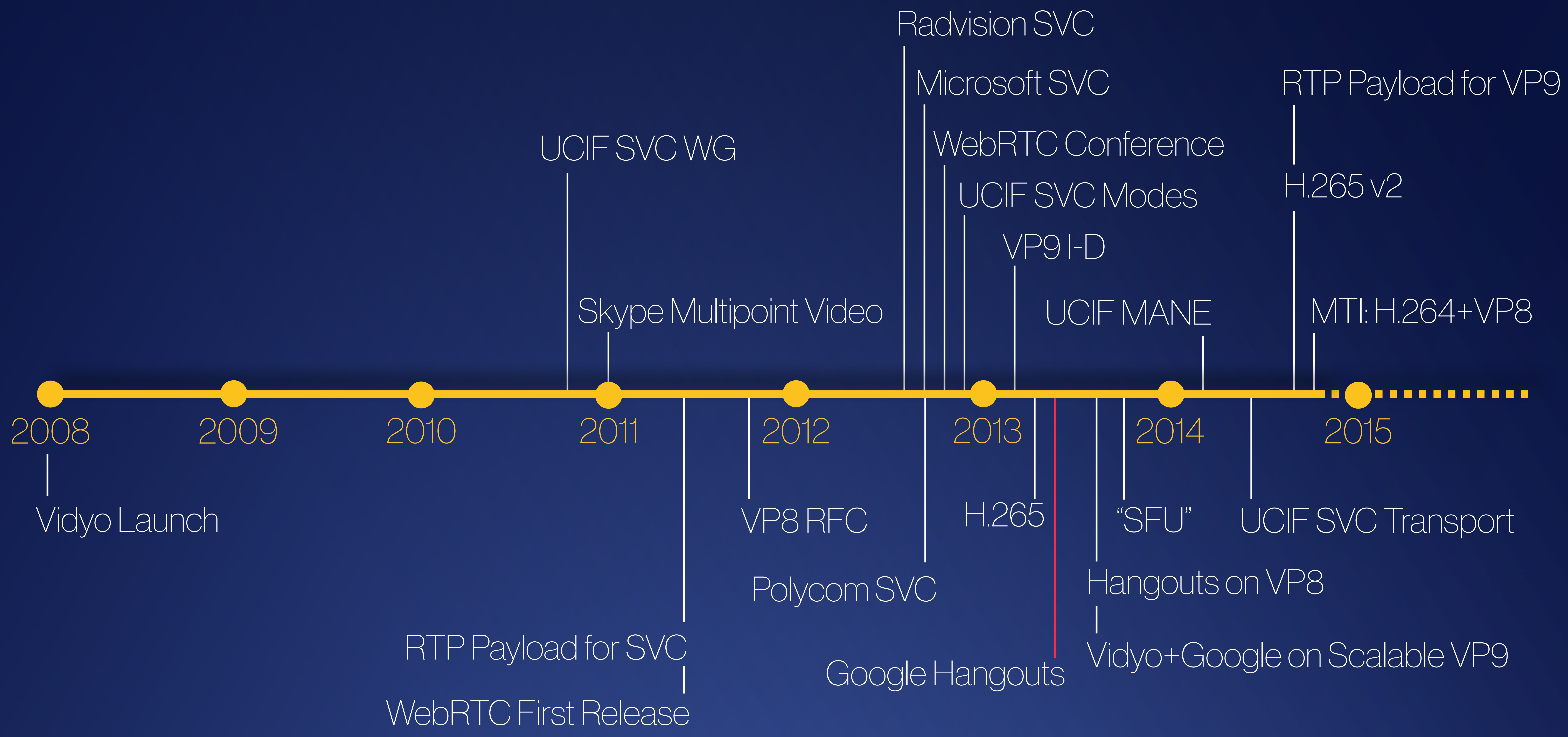
ITU-T
TELECOMMUNICATION STANDARDIZATION SECTOR OF ITU

H.265
(04/2013)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS
Infrastructure of audiovisual services – Coding of moving video

High efficiency video coding

Recommendation ITU-T H.265



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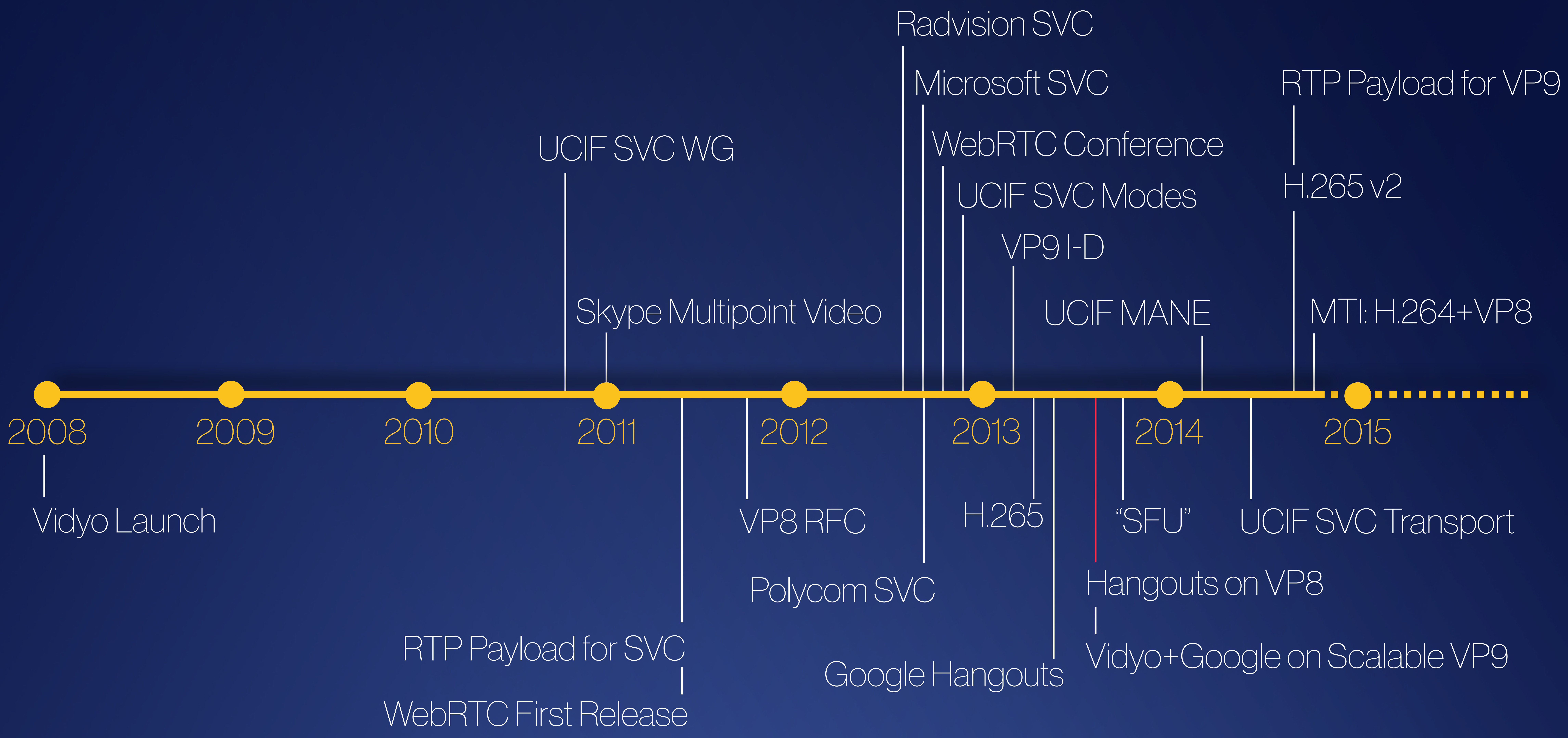
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

Google Hangouts

May 2013

Using H.264 SVC



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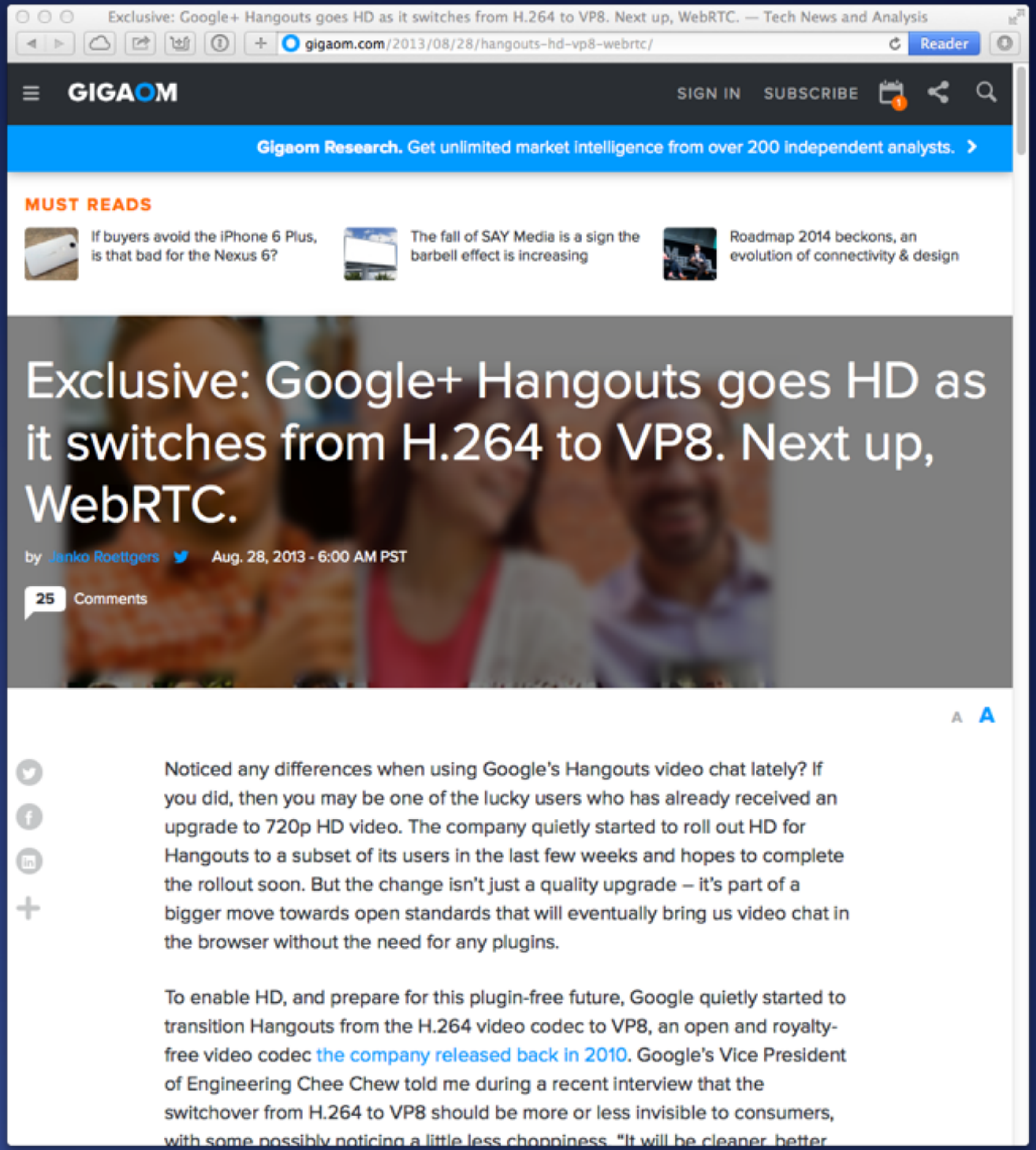
Chief Scientist & Co-founder

Hangouts on VP8

August 2013

Switches to VP8


Exclusive: Google+ Hangouts goes HD as it switches from H.264 to VP8. Next up, WebRTC. — Tech News and Analysis

gigaom.com/2013/08/28/hangouts-hd-vp8-webrtc/

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If buyers avoid the iPhone 6 Plus, is that bad for the Nexus 6?

The fall of SAY Media is a sign the barbell effect is increasing

Roadmap 2014 beckons, an evolution of connectivity & design

Exclusive: Google+ Hangouts goes HD as it switches from H.264 to VP8. Next up, WebRTC.

by Janko Roettgers Aug. 28, 2013 - 6:00 AM PST

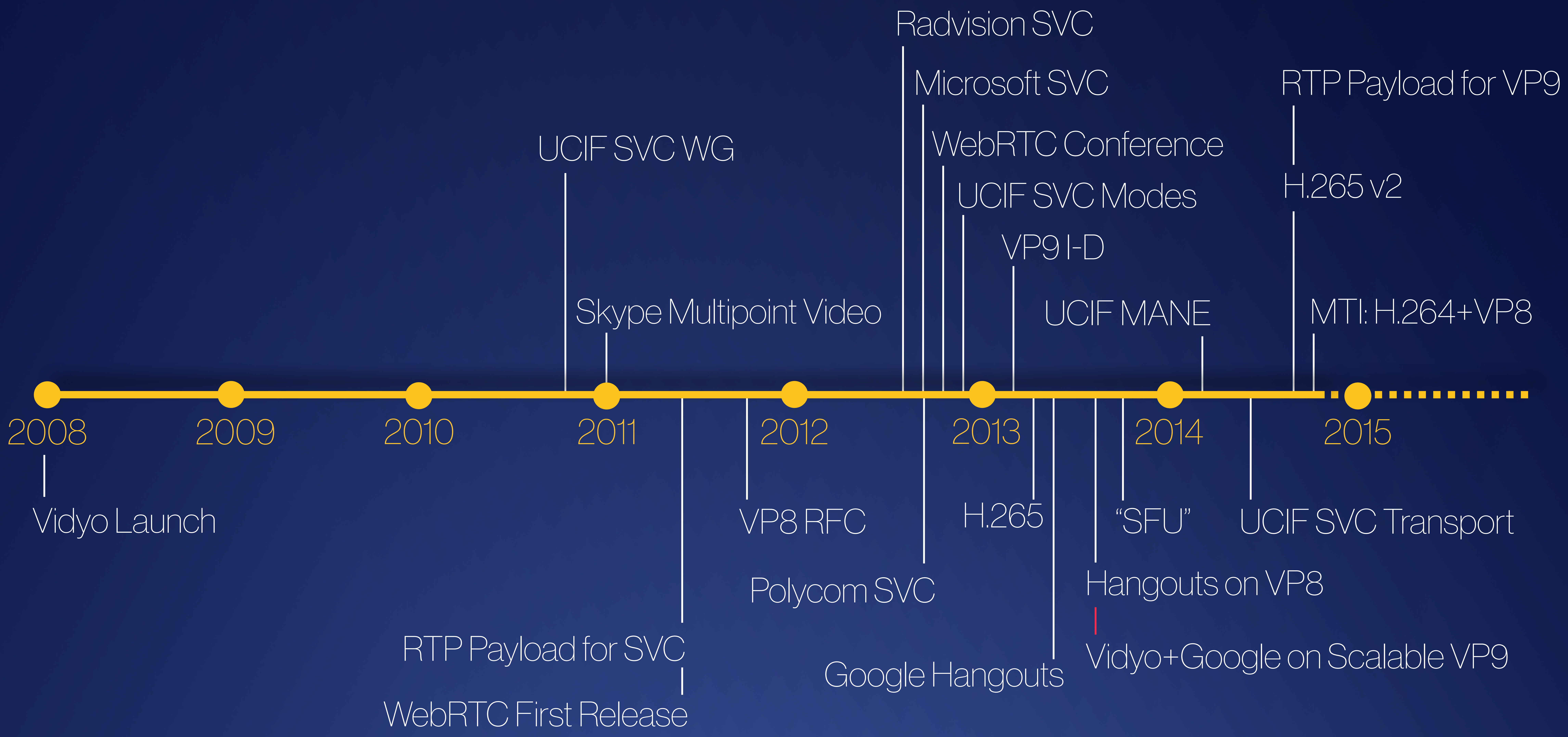
25 Comments

Noticed any differences when using Google's Hangouts video chat lately? If you did, then you may be one of the lucky users who has already received an upgrade to 720p HD video. The company quietly started to roll out HD for Hangouts to a subset of its users in the last few weeks and hopes to complete the rollout soon. But the change isn't just a quality upgrade – it's part of a bigger move towards open standards that will eventually bring us video chat in the browser without the need for any plugins.

To enable HD, and prepare for this plugin-free future, Google quietly started to transition Hangouts from the H.264 video codec to VP8, an open and royalty-free video codec the company released back in 2010. Google's Vice President of Engineering Chee Chew told me during a recent interview that the switchover from H.264 to VP8 should be more or less invisible to consumers, with some possibly noticing a little less choppiness. "It will be cleaner, better

Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



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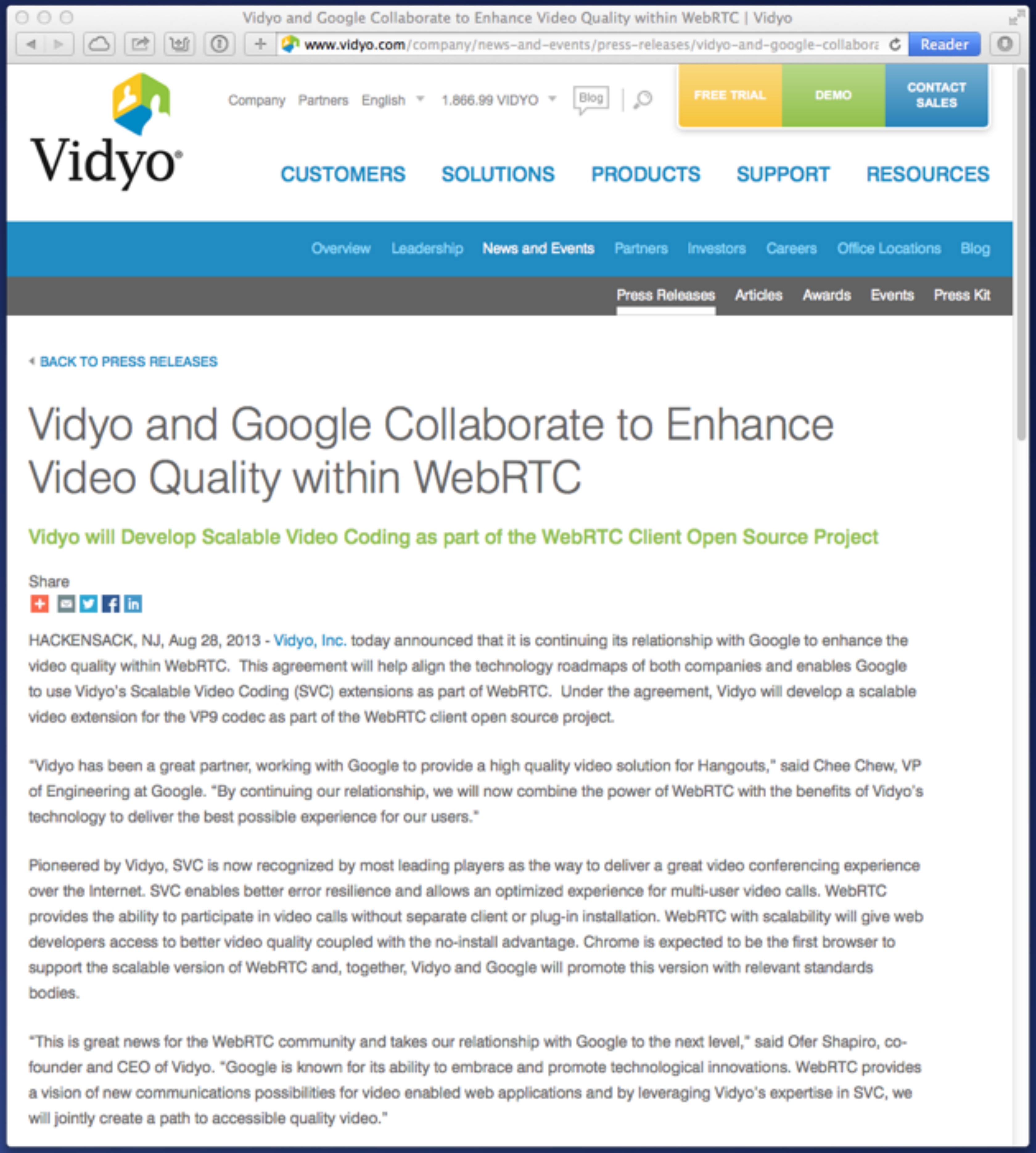
Chief Scientist & Co-founder

Vidyo+Google on Scalable VP9

August 2013

Scalable Video Coding for VP9 in WebRTC



Vidyo and Google Collaborate to Enhance Video Quality within WebRTC | Vidyo

www.vidyo.com/company/news-and-events/press-releases/vidyo-and-google-collabora

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Vidyo and Google Collaborate to Enhance Video Quality within WebRTC

Vidyo will Develop Scalable Video Coding as part of the WebRTC Client Open Source Project

Share

HACKENSACK, NJ, Aug 28, 2013 - Vidyo, Inc. today announced that it is continuing its relationship with Google to enhance the video quality within WebRTC. This agreement will help align the technology roadmaps of both companies and enables Google to use Vidyo's Scalable Video Coding (SVC) extensions as part of WebRTC. Under the agreement, Vidyo will develop a scalable video extension for the VP9 codec as part of the WebRTC client open source project.

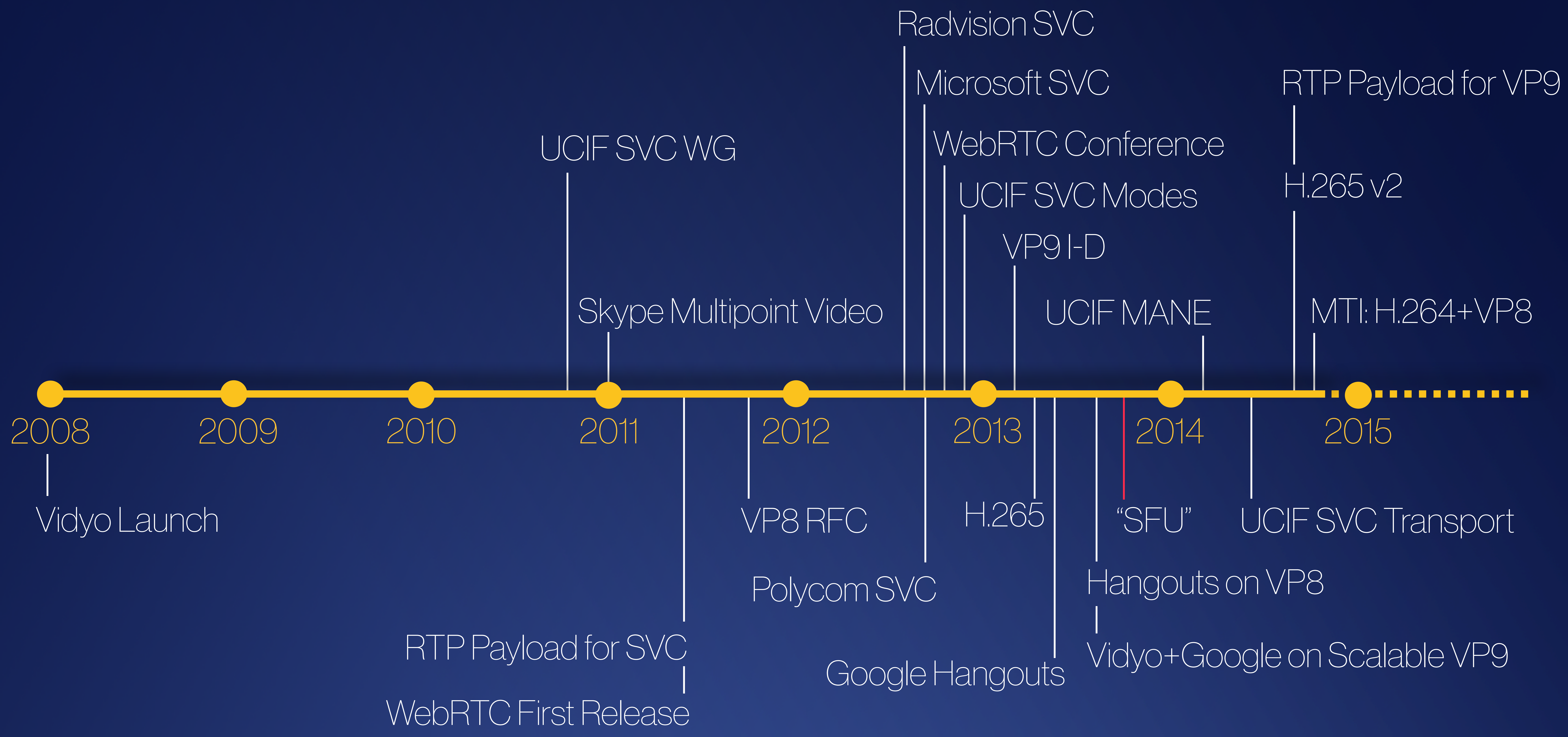
"Vidyo has been a great partner, working with Google to provide a high quality video solution for Hangouts," said Chee Chew, VP of Engineering at Google. "By continuing our relationship, we will now combine the power of WebRTC with the benefits of Vidyo's technology to deliver the best possible experience for our users."

Pioneered by Vidyo, SVC is now recognized by most leading players as the way to deliver a great video conferencing experience over the Internet. SVC enables better error resilience and allows an optimized experience for multi-user video calls. WebRTC provides the ability to participate in video calls without separate client or plug-in installation. WebRTC with scalability will give web developers access to better video quality coupled with the no-install advantage. Chrome is expected to be the first browser to support the scalable version of WebRTC and, together, Vidyo and Google will promote this version with relevant standards bodies.

"This is great news for the WebRTC community and takes our relationship with Google to the next level," said Ofer Shapiro, co-founder and CEO of Vidyo. "Google is known for its ability to embrace and promote technological innovations. WebRTC provides a vision of new communications possibilities for video enabled web applications and by leveraging Vidyo's expertise in SVC, we will jointly create a path to accessible quality video."

Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder



Vidyo

Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

"SFU"

October 2013
Westerlund and Wenger
draft-ietf-avtcore-rtp-topologies-update-01

Introduction of the term
"Selective Forwarding Unit" - SFU

```
Network Working Group                                      M. Westerlund
Internet-Draft                                                  Ericsson
Obsoletes: 5117 (if approved)                                  S. Wenger
Intended status: Informational                                     Vidyo
Expires: April 25, 2014                                 October 22, 2013


                             RTP Topologies
               draft-ietf-avtcore-rtp-topologies-update-01

Abstract

   This document discusses point to point and multi-endpoint topologies
   used in Real-time Transport Protocol (RTP)-based environments.  In
   particular, centralized topologies commonly employed in the video
   conferencing industry are mapped to the RTP terminology.

   This document is updated with additional topologies and is intended
   to replace RFC 5117.

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Westerlund & Wenger      Expires April 25, 2014                 [Page 1]
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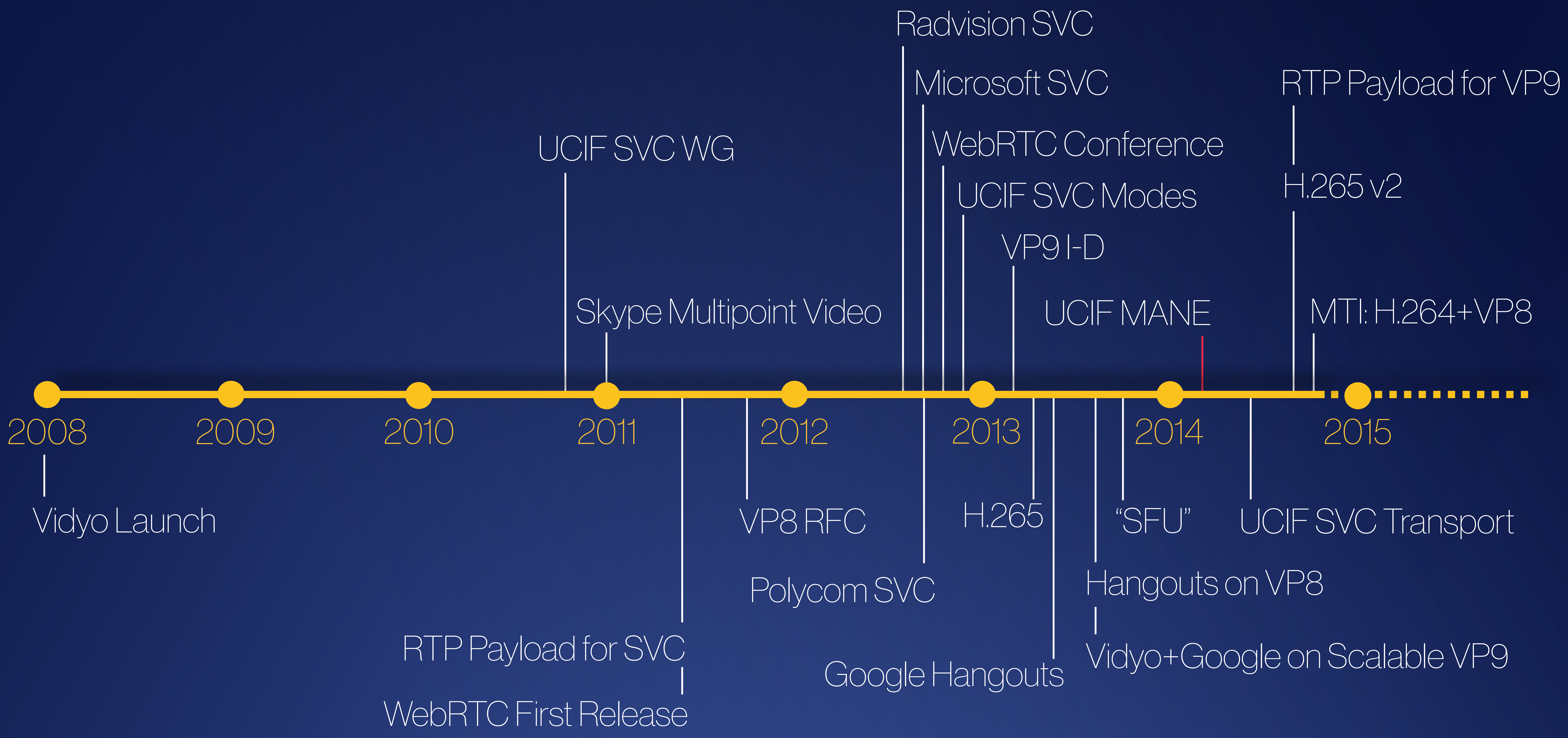

Vidyo

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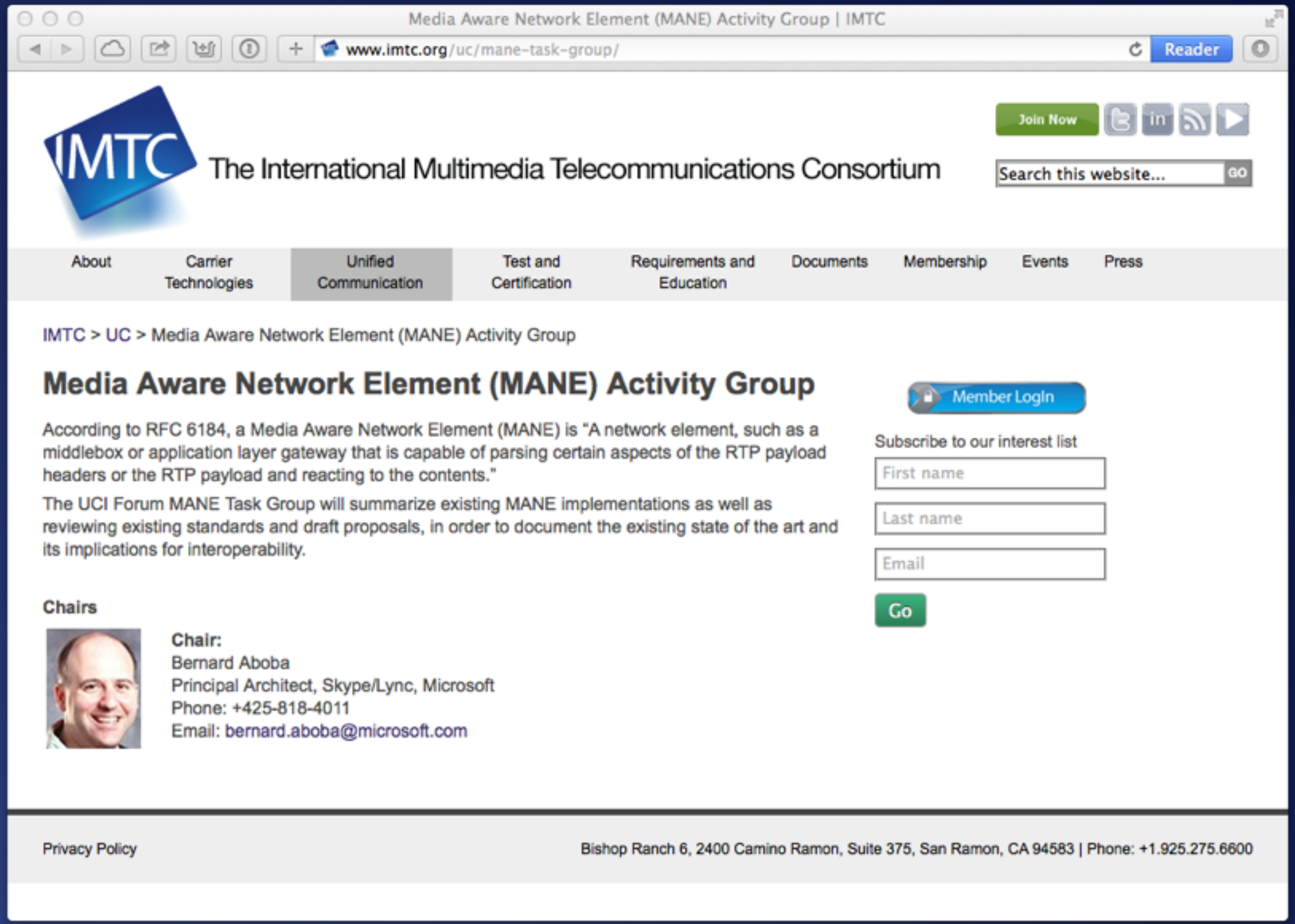
Chief Scientist & Co-founder

UCIF MANE

February 2014

Investigate SFU implementations, as well as existing standards and draft proposals



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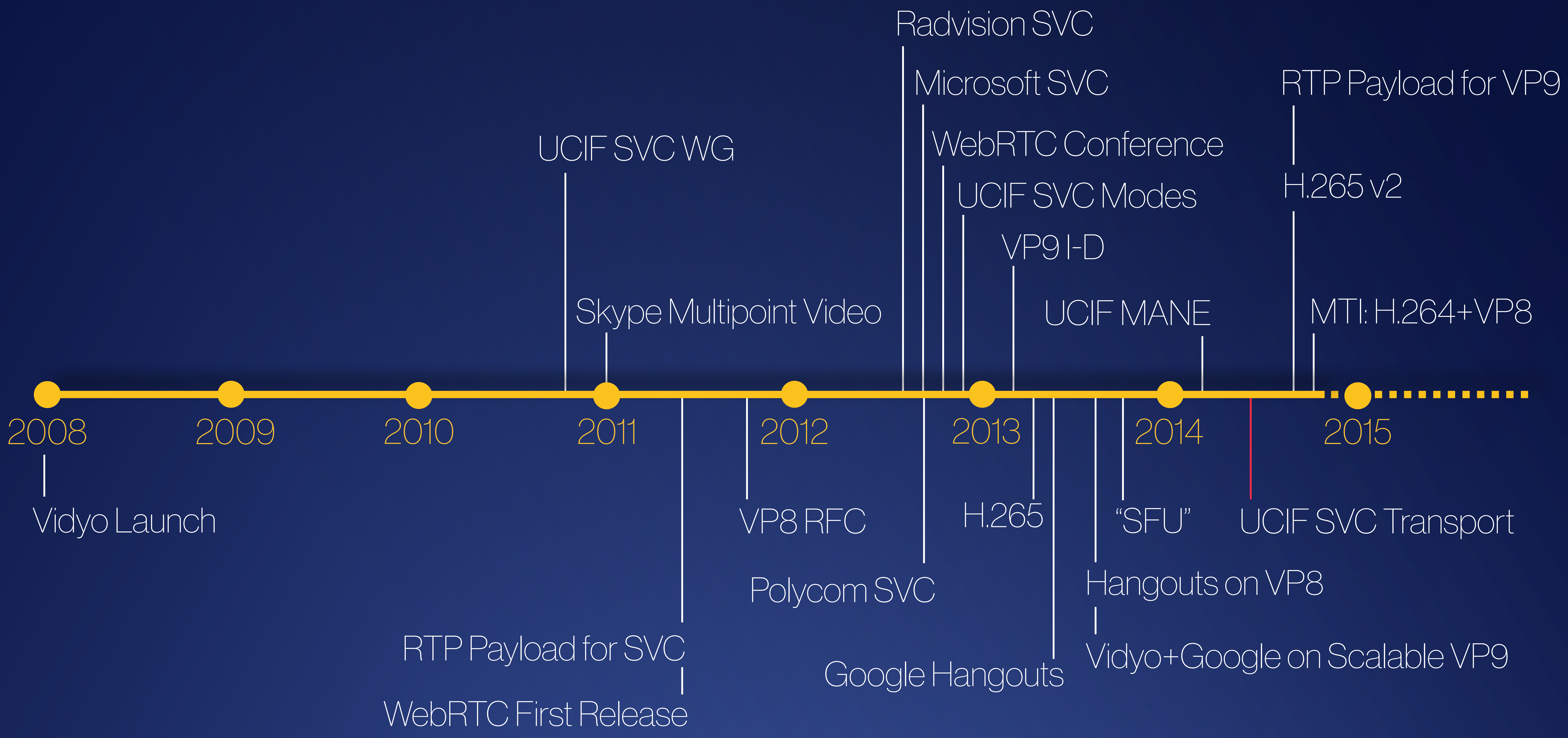

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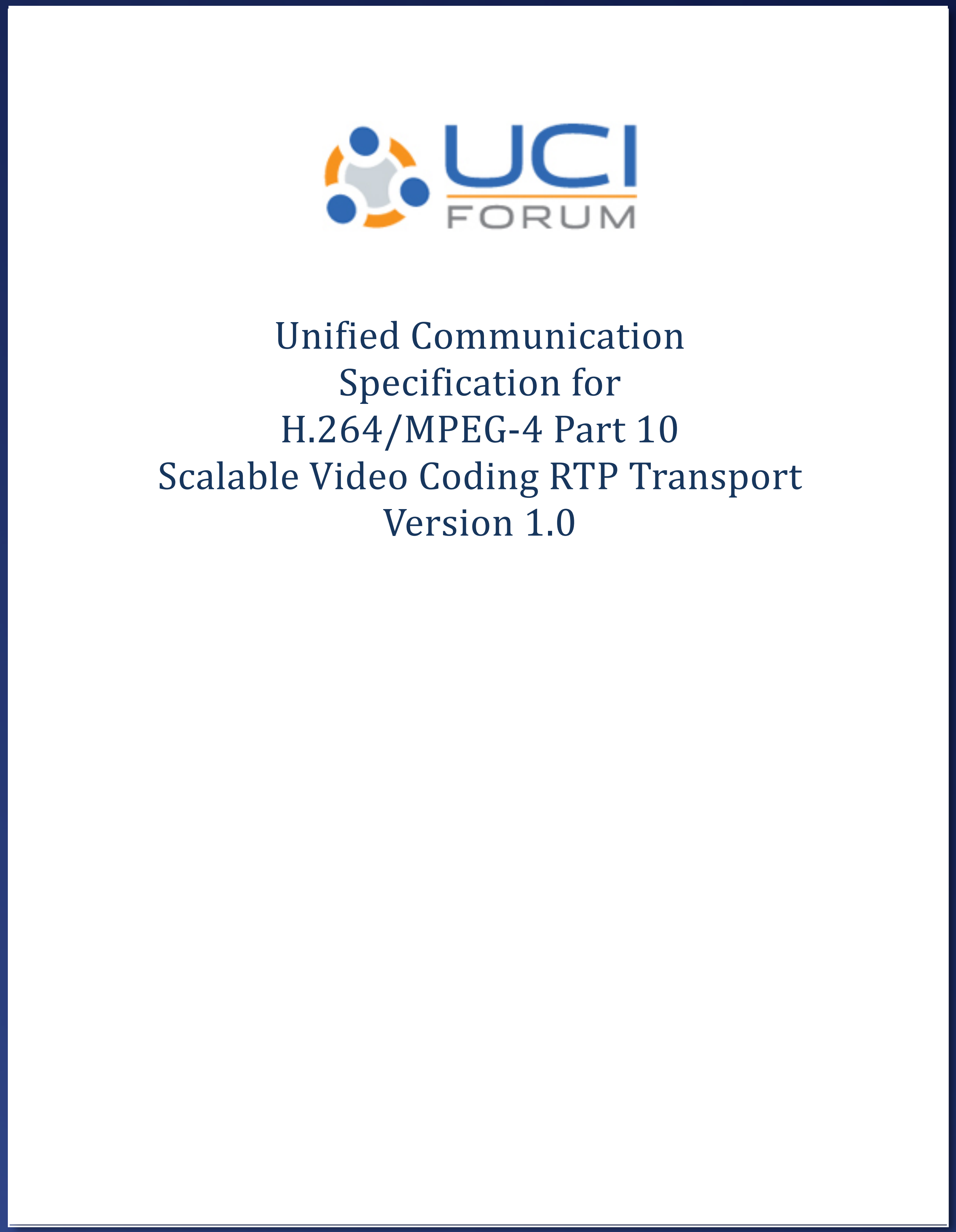
UCIF SVC Transport

June 2014

SVC RTP Transport


UCI FORUM

Unified Communication
Specification for
H.264/MPEG-4 Part 10
Scalable Video Coding RTP Transport
Version 1.0

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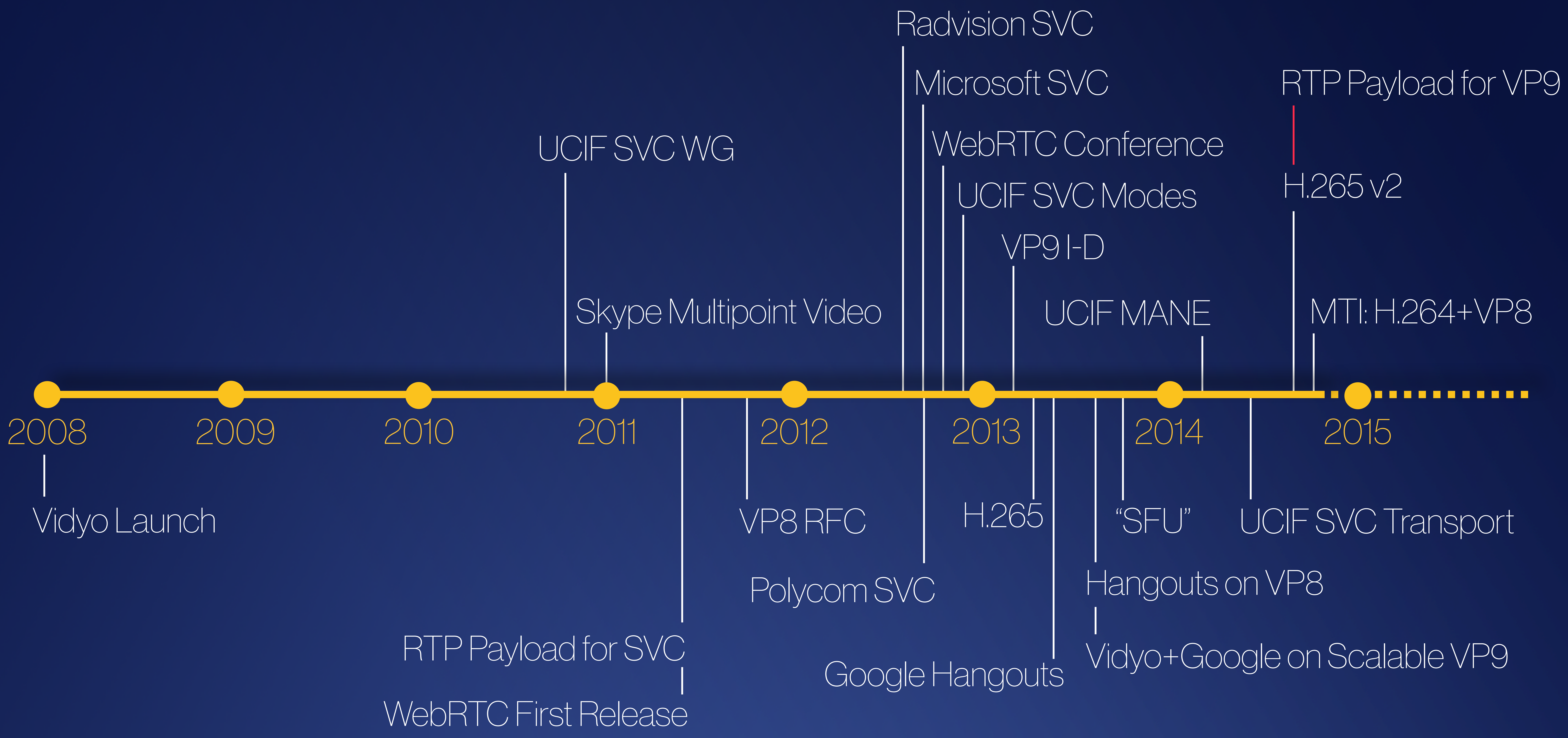

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RTP Payload for VP9

October 2014
Uberti, Holmer, Flodman, and Lennox
draft-uberti-payload-vp9

How to packetize VP9 video for RTP transport, plus signaling parameters

Payload Working Group — J. Uberti
Internet-Draft — S. Holmer
Intended status: Standards Track — M. Flodman
Expires: April 30, 2015 — Google
J. Lennox
Vidyo
October 27, 2014

RTP Payload Format for VP9 Video
draft-uberti-payload-vp9-00

Abstract

This memo describes an RTP payload format for the VP9 video codec. The payload format has wide applicability, as it supports applications from low bit-rate peer-to-peer usage, to high bit-rate video conferences. It includes provisions for temporal and spatial scalability.

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Uberti, et al. — Expires April 30, 2015 — [Page 1]

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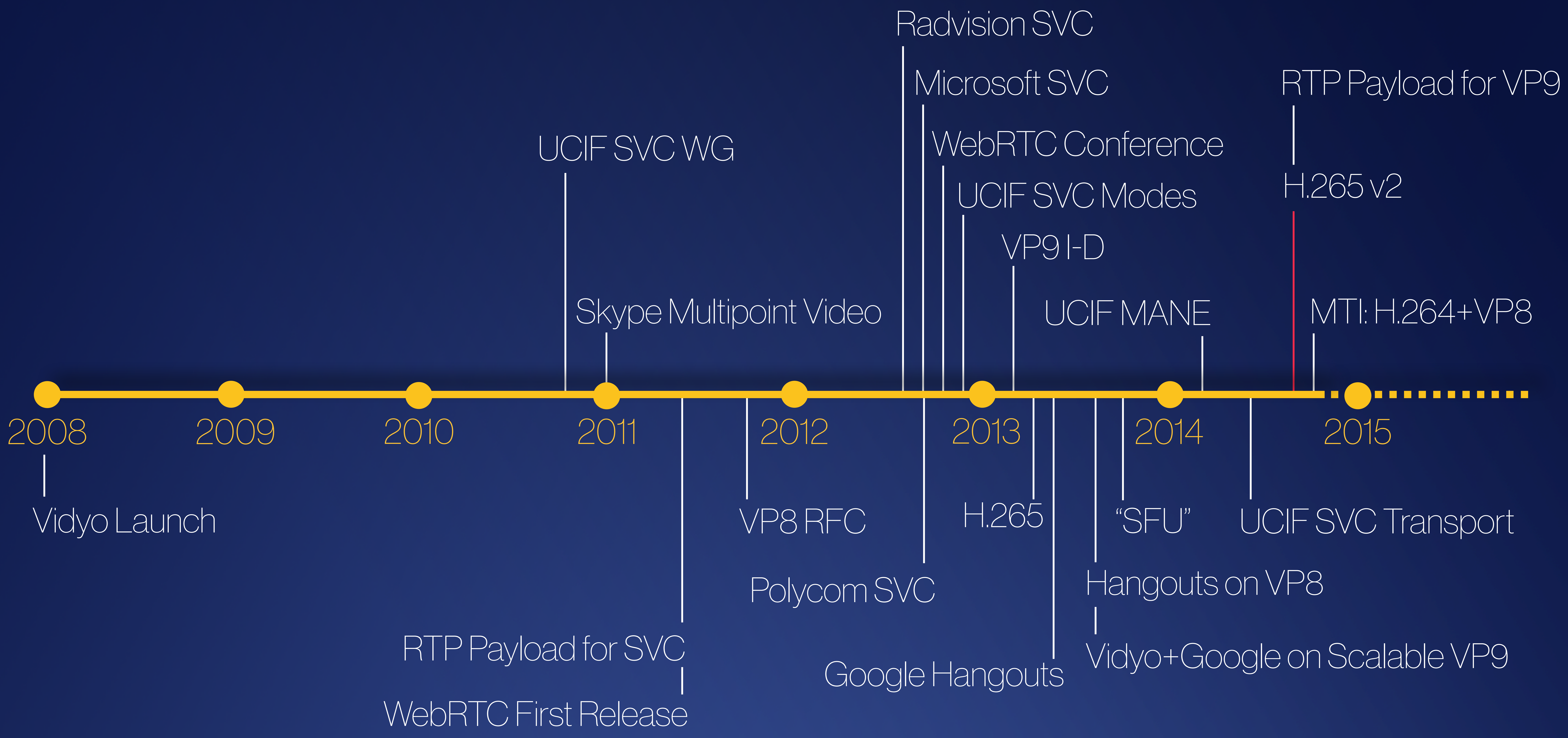


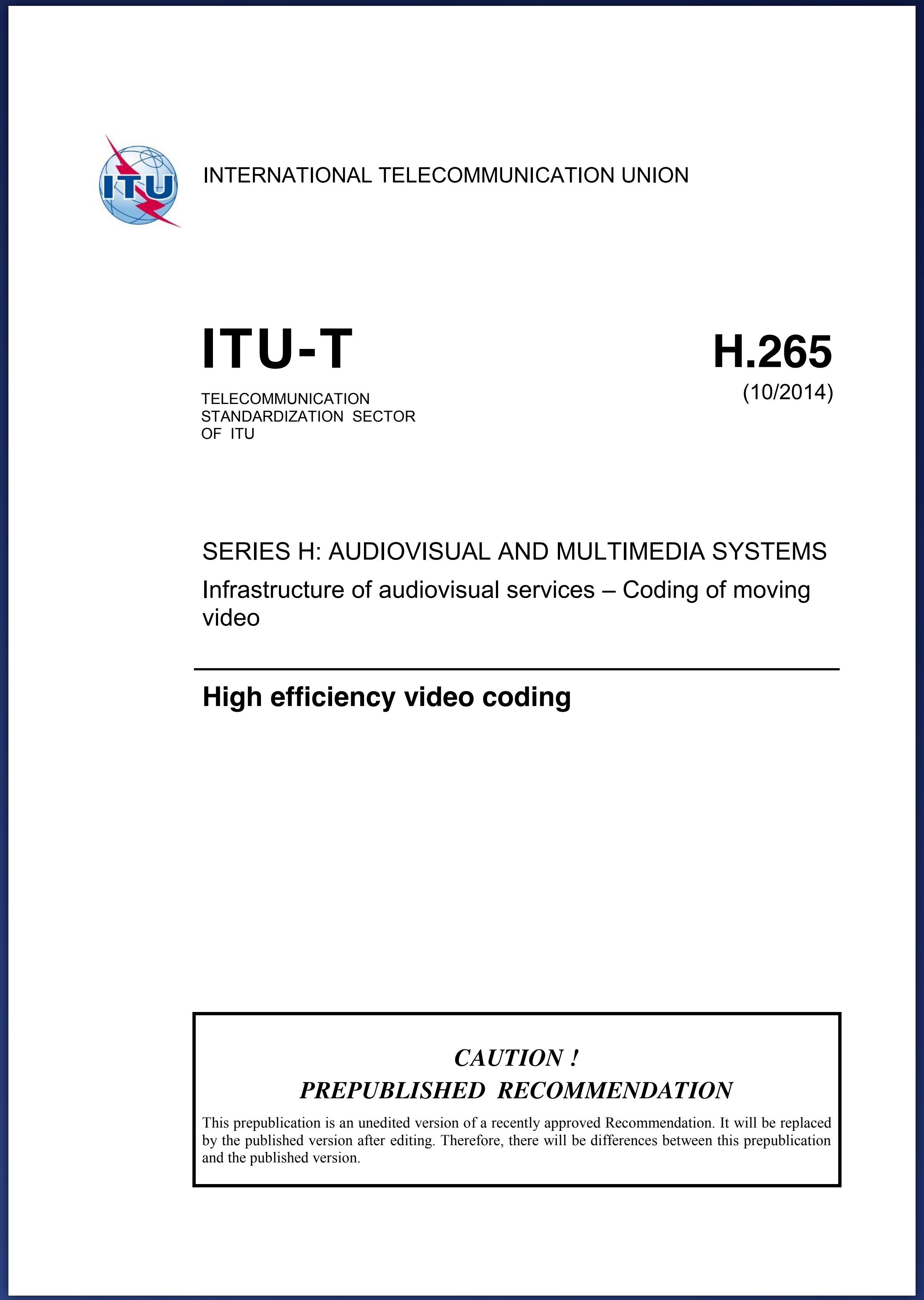
Vidyo
Alex Eleftheriadis, Ph. D.
Chief Scientist & Co-founder

H.265 v2

October 2014

HEVC version 2

Spatial and quality scalability extensions (SHVC)



INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION STANDARDIZATION SECTOR OF ITU

H.265

(10/2014)

SERIES H: AUDIOVISUAL AND MULTIMEDIA SYSTEMS

Infrastructure of audiovisual services – Coding of moving video

High efficiency video coding

CAUTION !

PREPUBLISHED RECOMMENDATION

This prepublication is an unedited version of a recently approved Recommendation. It will be replaced by the published version after editing. Therefore, there will be differences between this prepublication and the published version.

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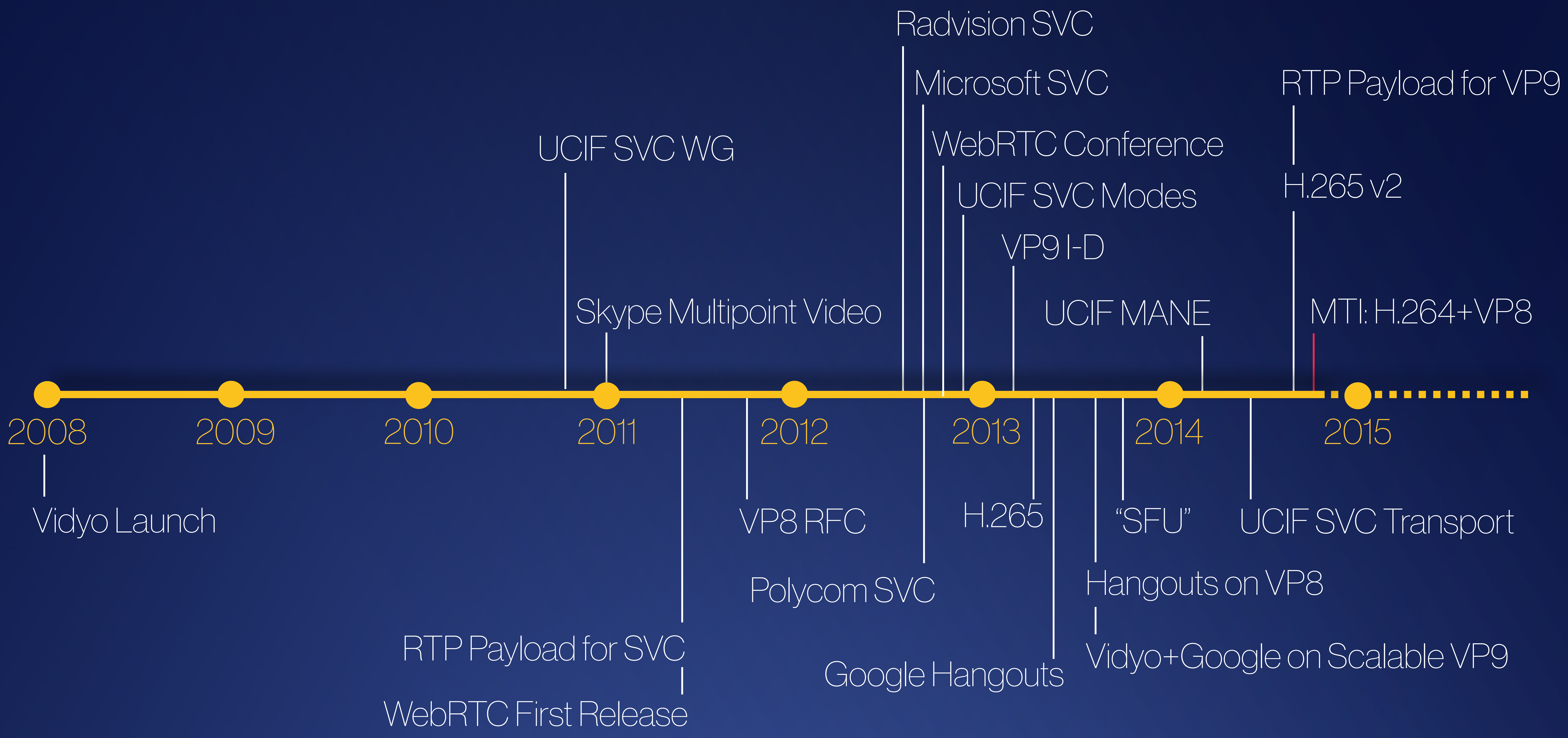
Alex Eleftheriadis, Ph. D.

Chief Scientist & Co-founder



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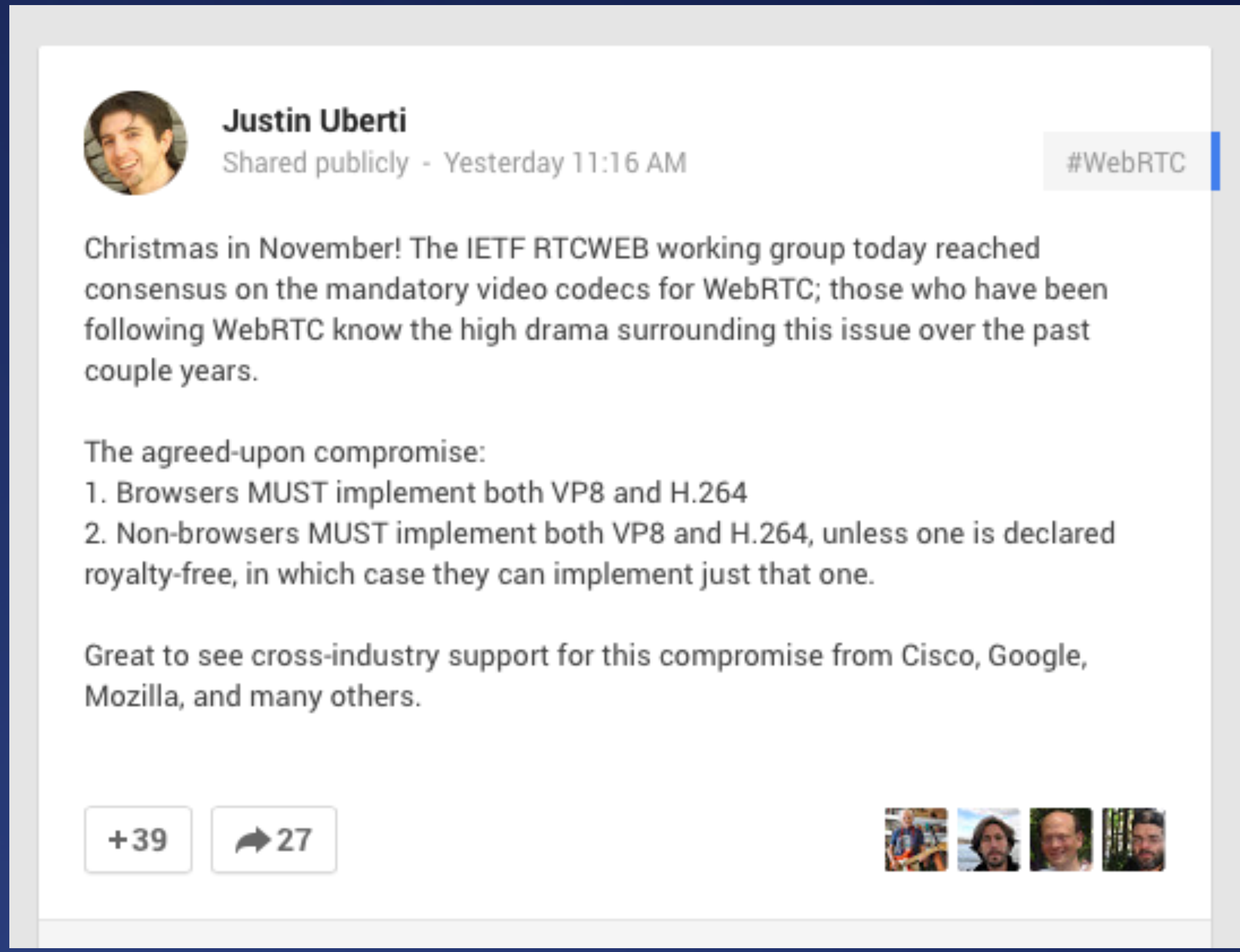
MTI: H.264+VP8

November 13, 2014

Mandatory-to-Implement Codec:
both H.264 and VP8



Justin Uberti
Shared publicly - Yesterday 11:16 AM
#WebRTC

Christmas in November! The IETF RTCWEB working group today reached consensus on the mandatory video codecs for WebRTC; those who have been following WebRTC know the high drama surrounding this issue over the past couple years.

The agreed-upon compromise:
1. Browsers MUST implement both VP8 and H.264
2. Non-browsers MUST implement both VP8 and H.264, unless one is declared royalty-free, in which case they can implement just that one.

Great to see cross-industry support for this compromise from Cisco, Google, Mozilla, and many others.

+39 27

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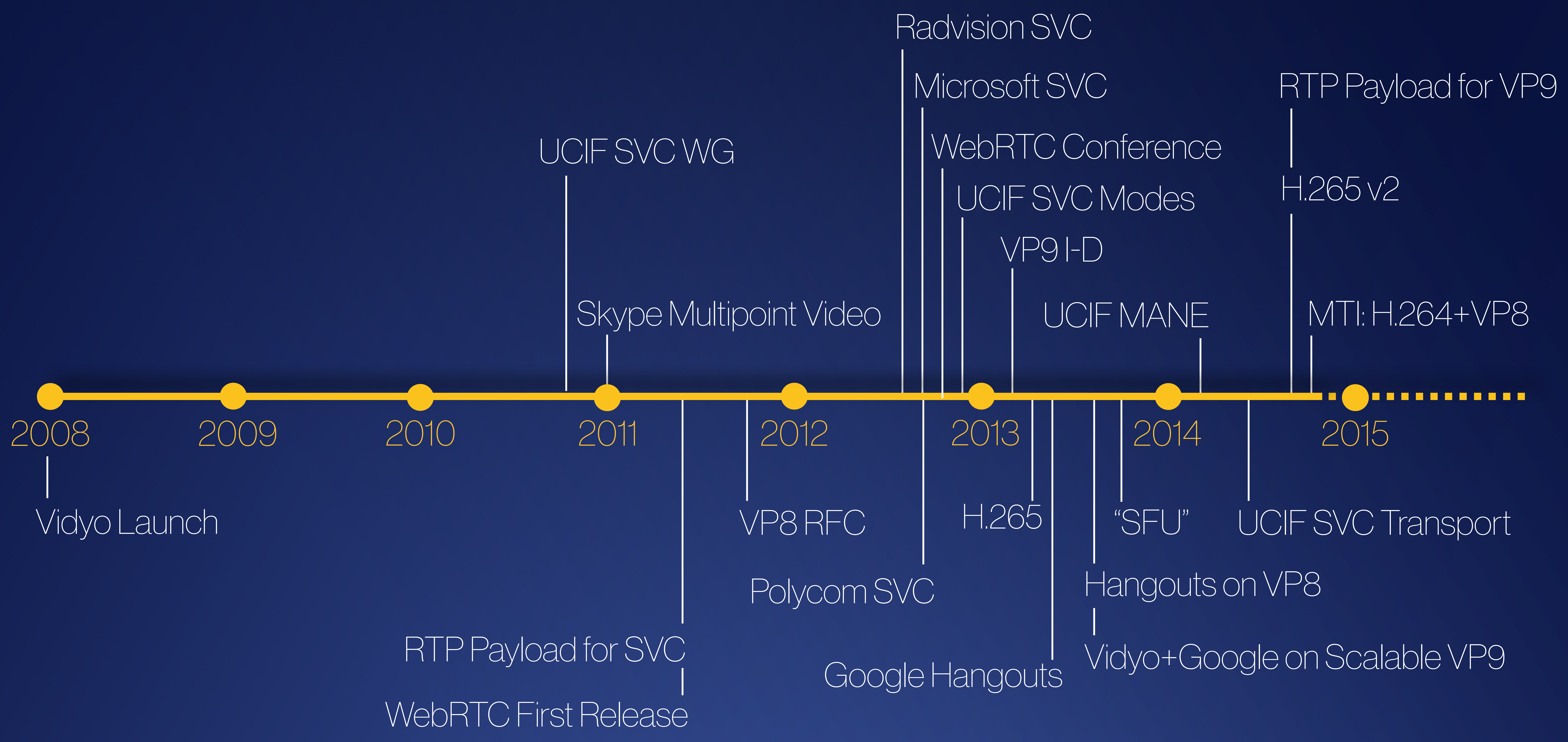

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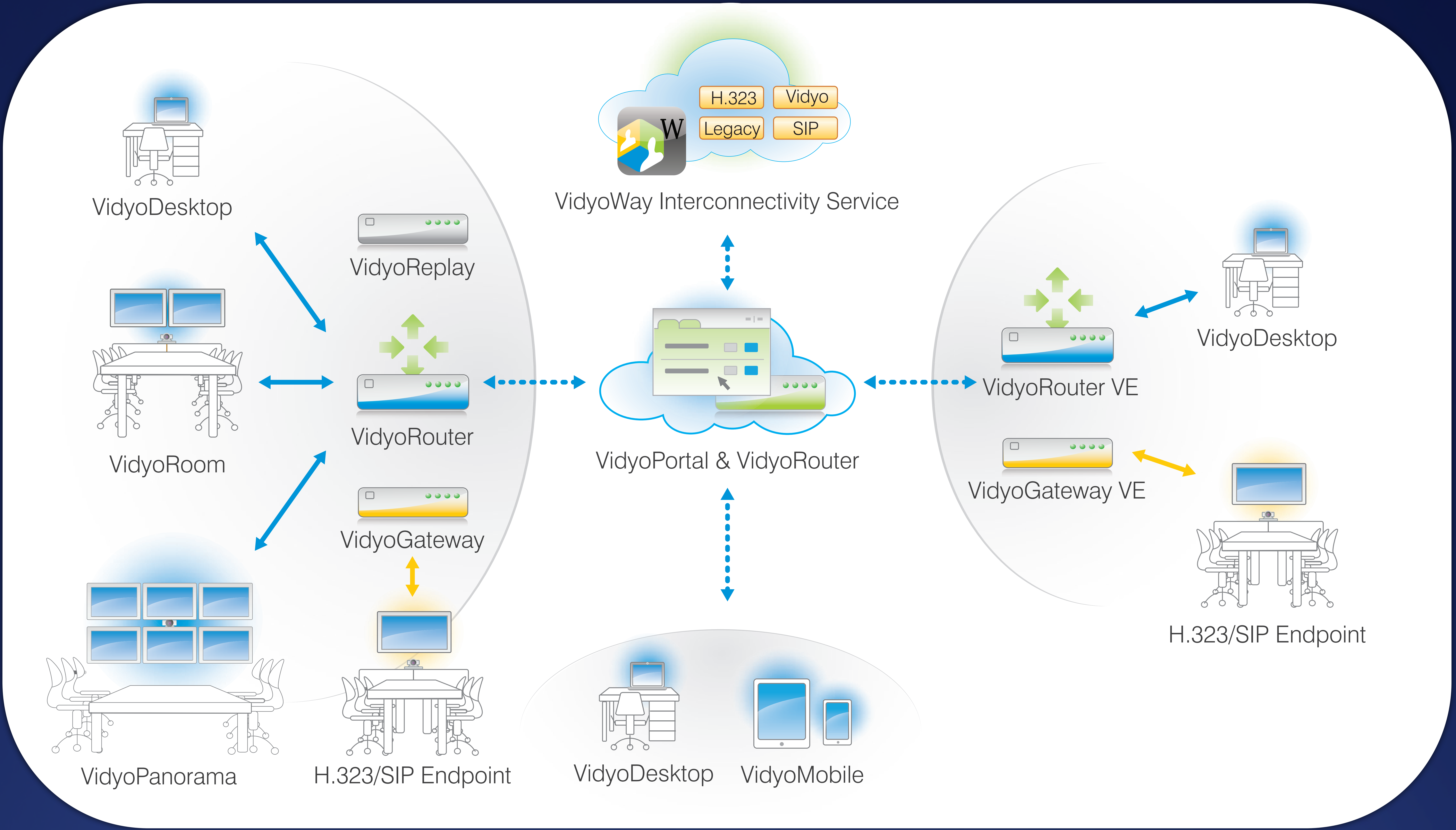
VidyoConferencing™ Portfolio



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© CERN

Vidyo at CERN

~18,000 users

800 simultaneous connections

up to 168 simultaneous H.323/SIP

12 phone access points worldwide

2 Vidyo Panoramas

(CERN site)

12 simultaneous recordings

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CERN Vidyo Worldwide Service Topology

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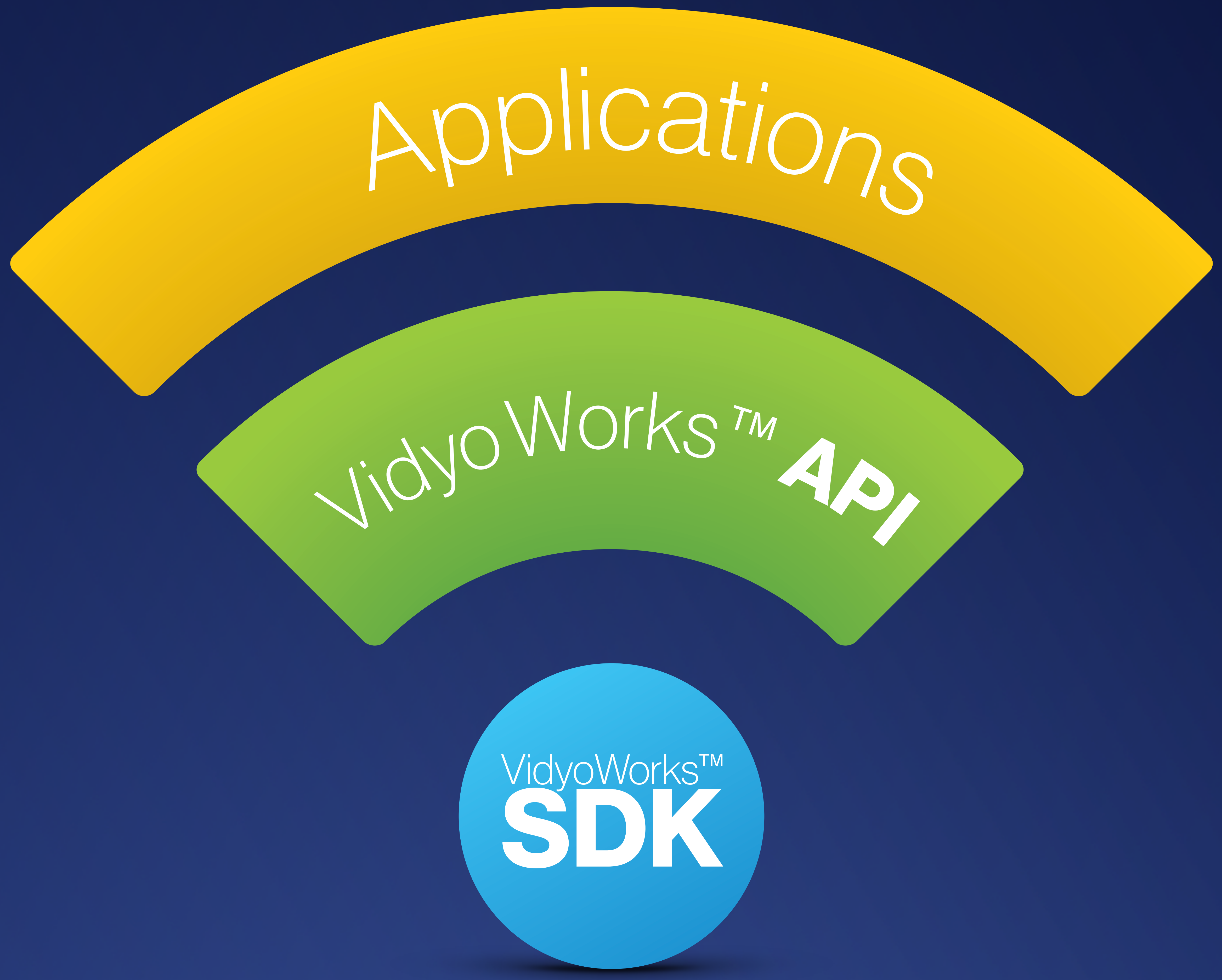

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WebRTC & VidyoWorks

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VidyoWorks™
Platform
router | portal | gateway | APIs

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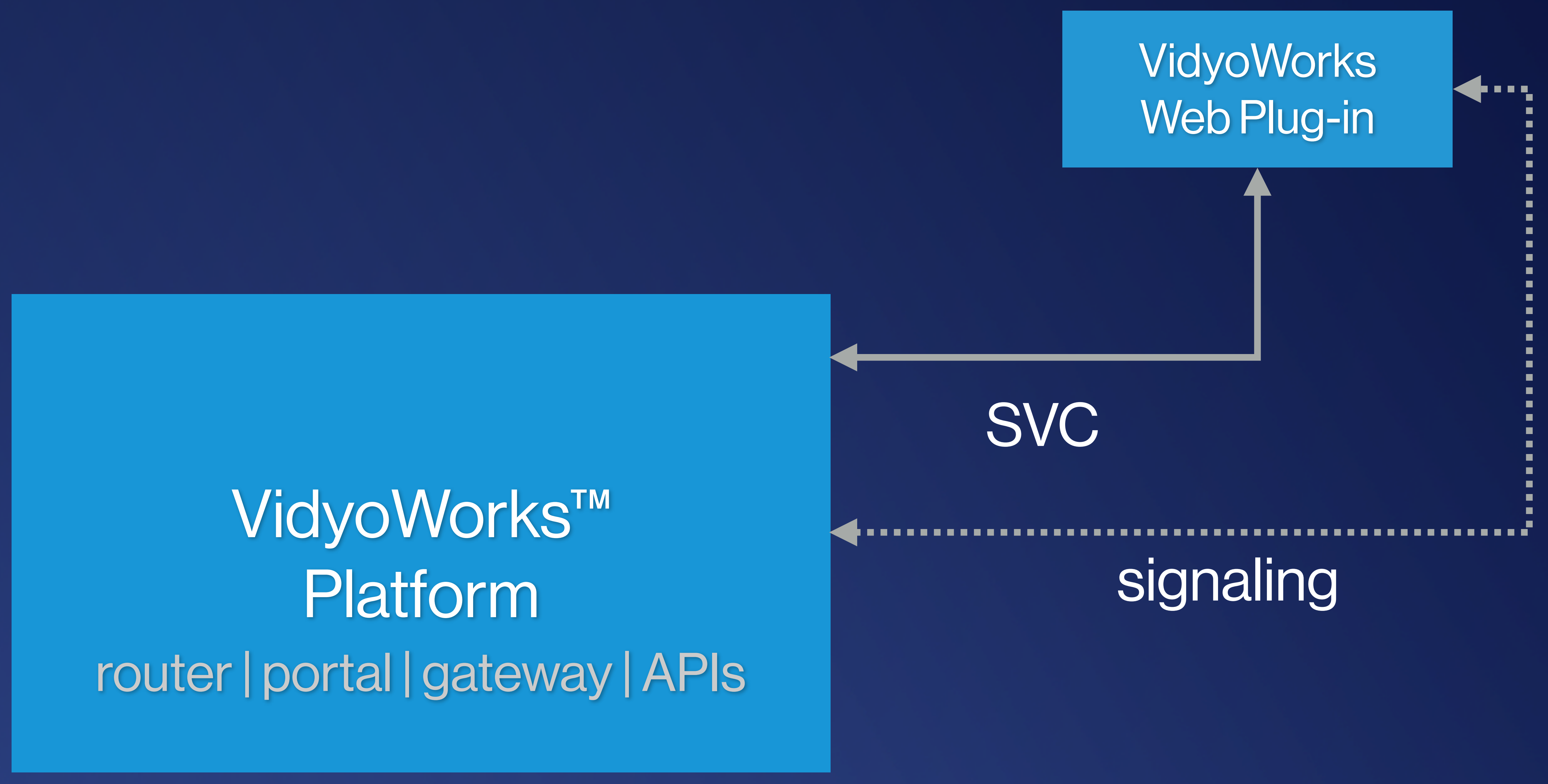
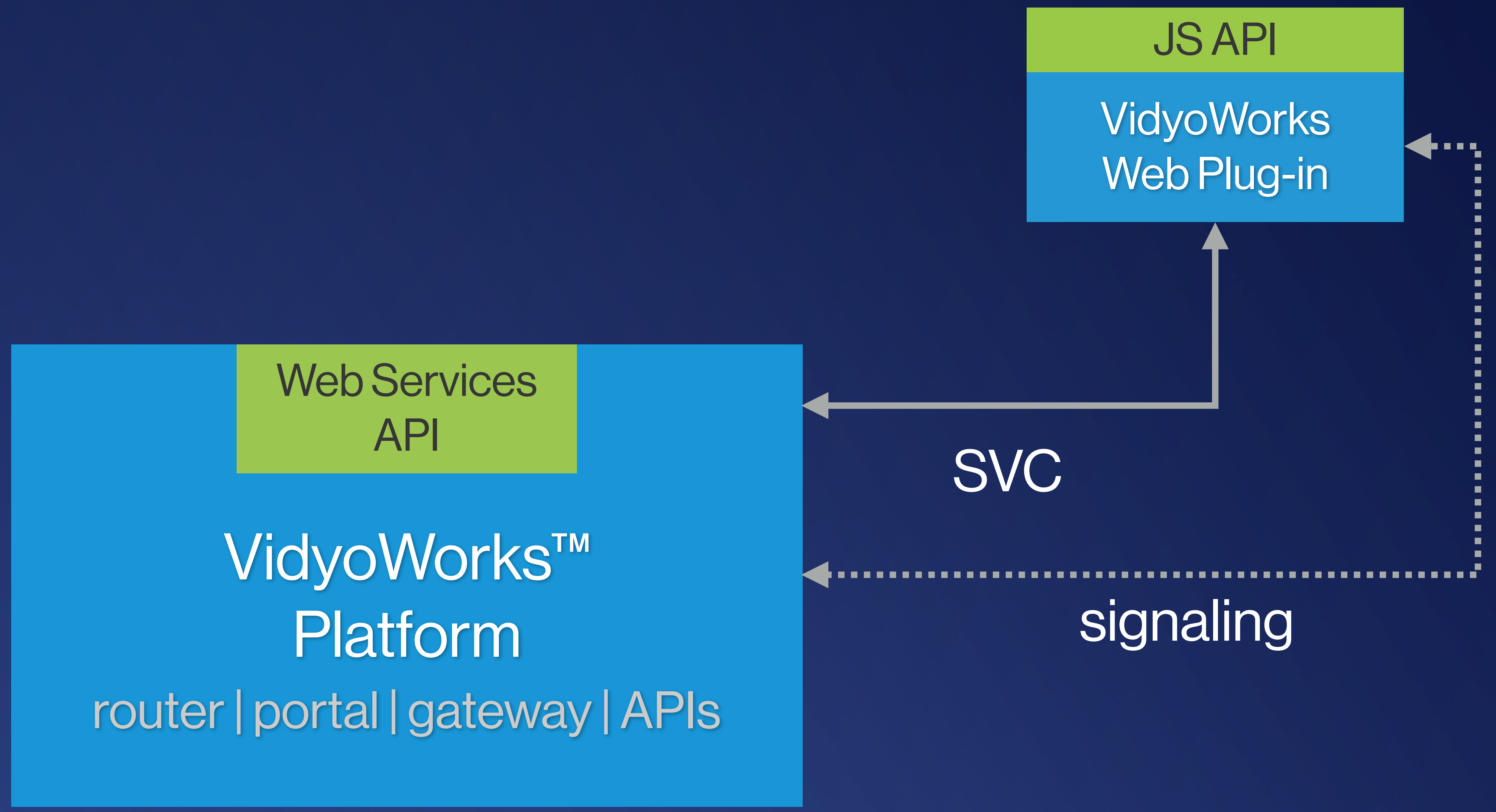
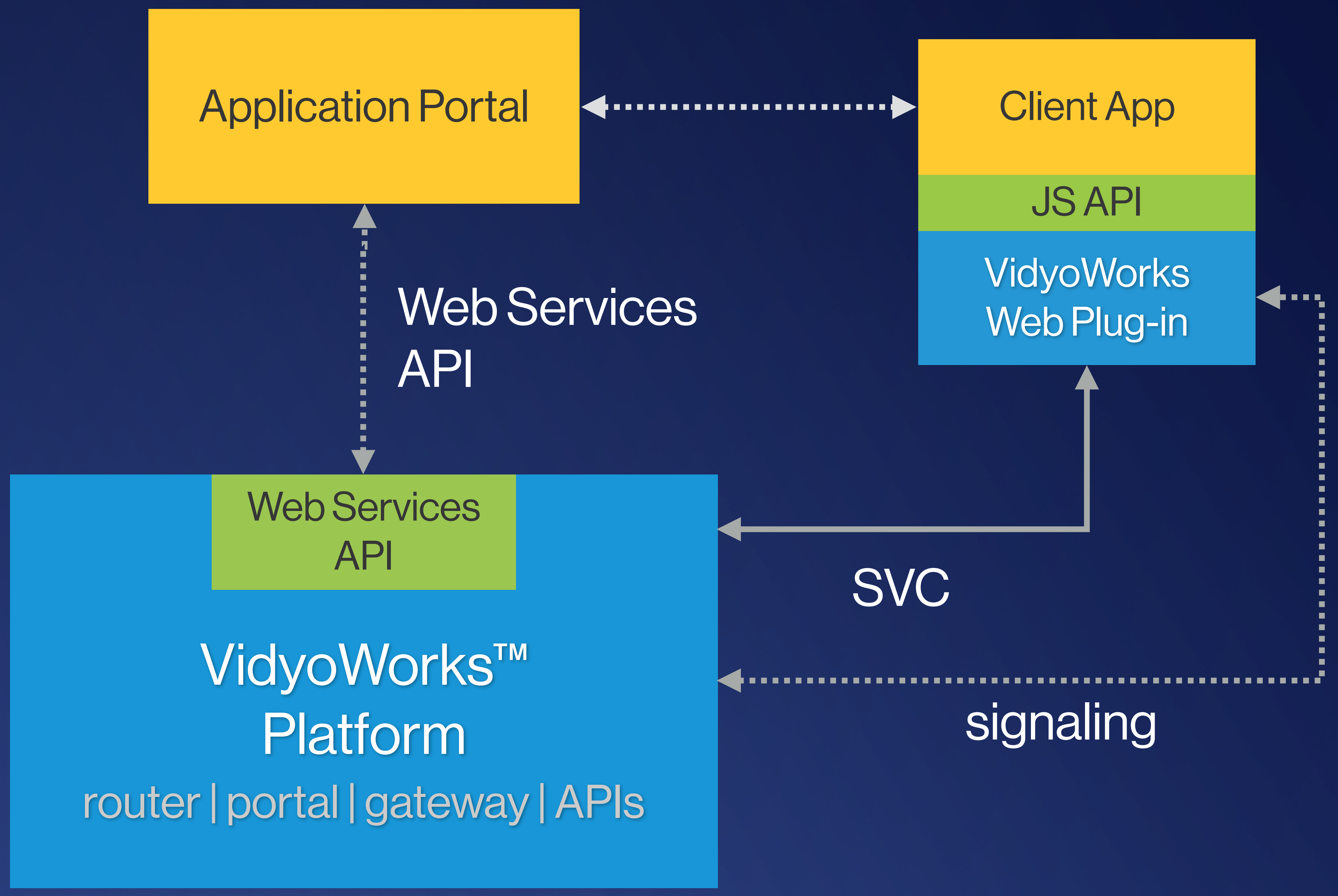

VidyoWorks
Web Plug-in
SVC
signaling
VidyoWorks™
Platform
router | portal | gateway | APIs


JS API
VidyoWorks
Web Plug-in
Web Services
API
VidyoWorks™
Platform
router | portal | gateway | APIs
SVC
signaling



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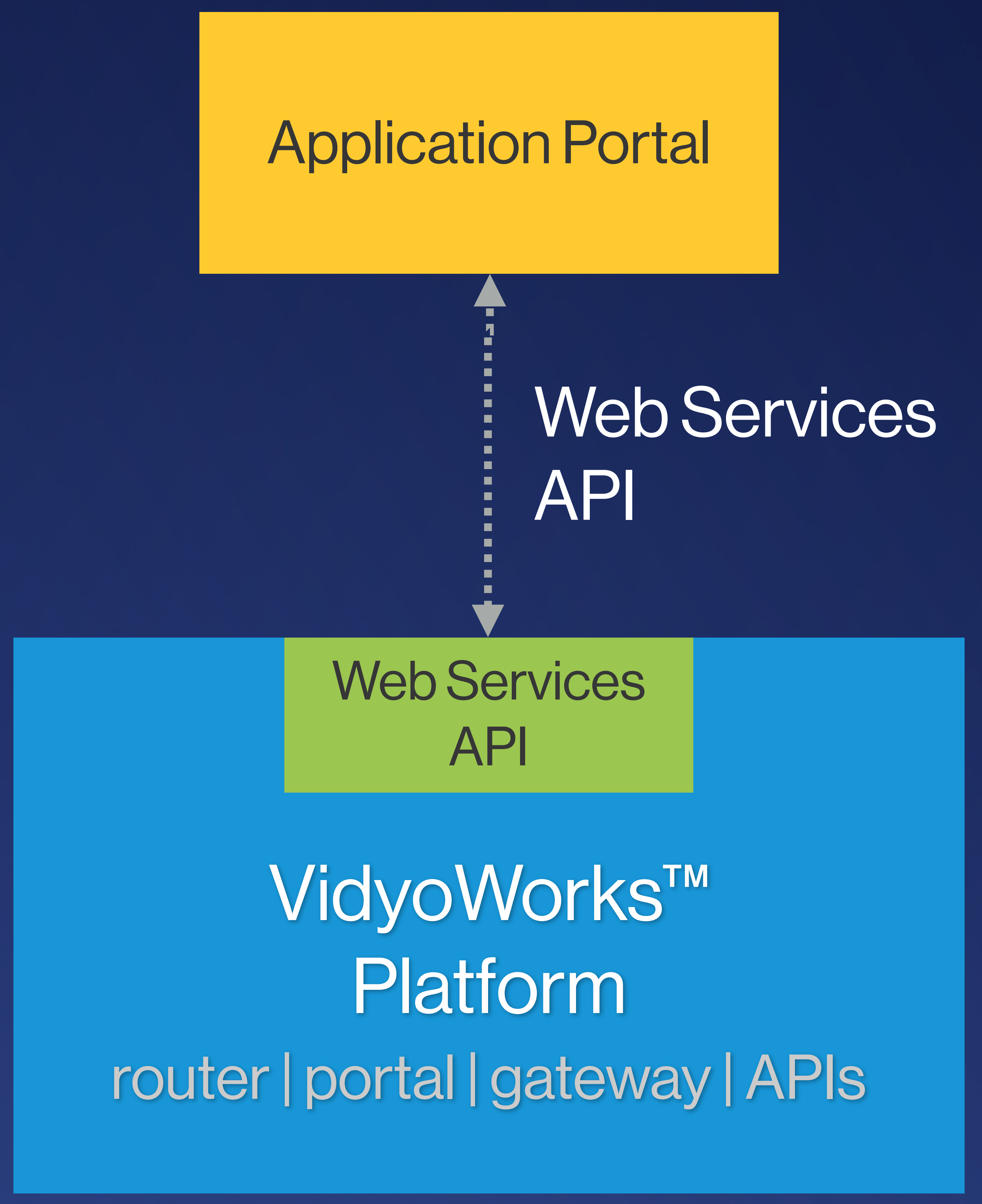
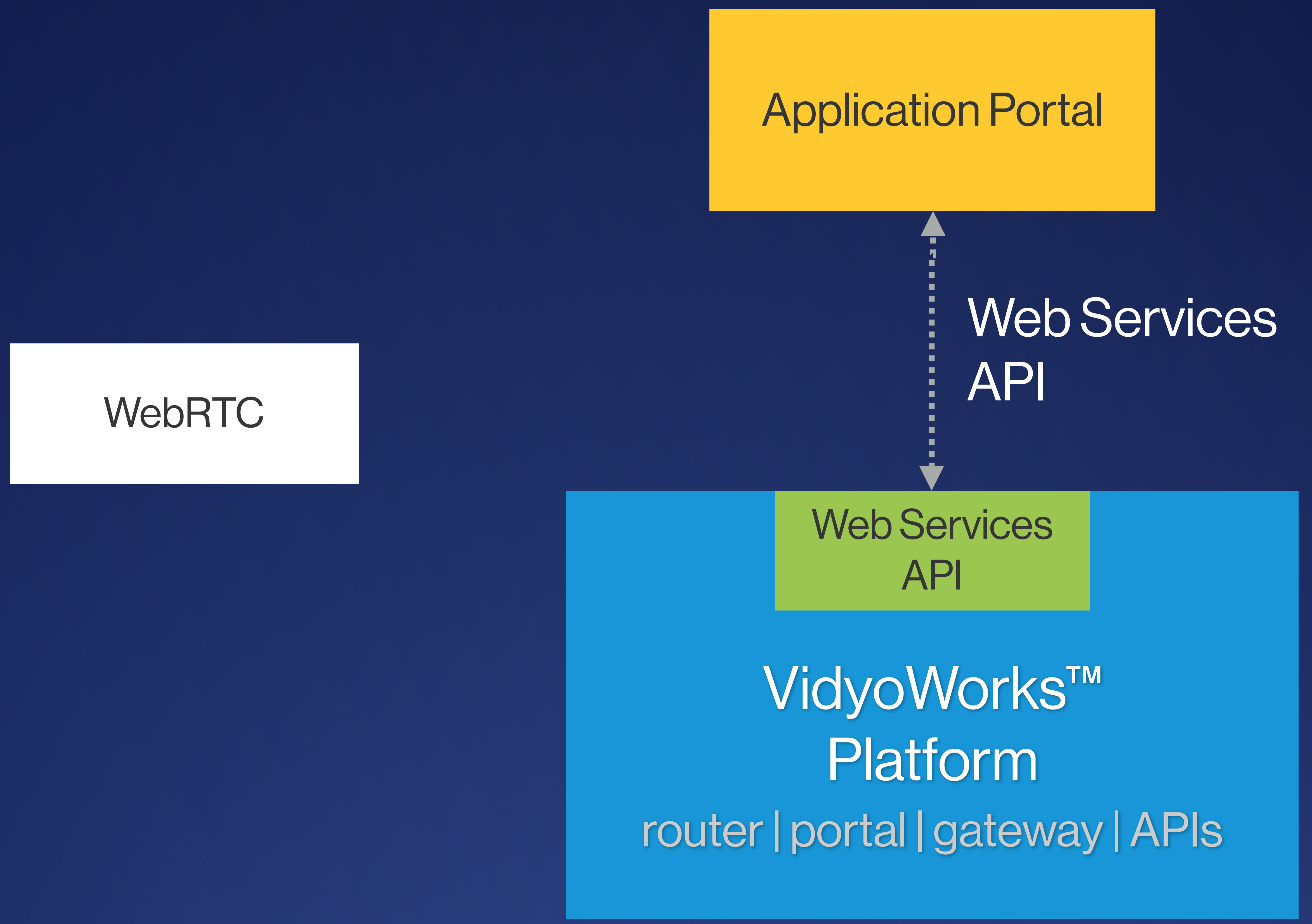

Application Portal
Web Services API
Web Services API
VidyoWorks™ Platform
router | portal | gateway | APIs



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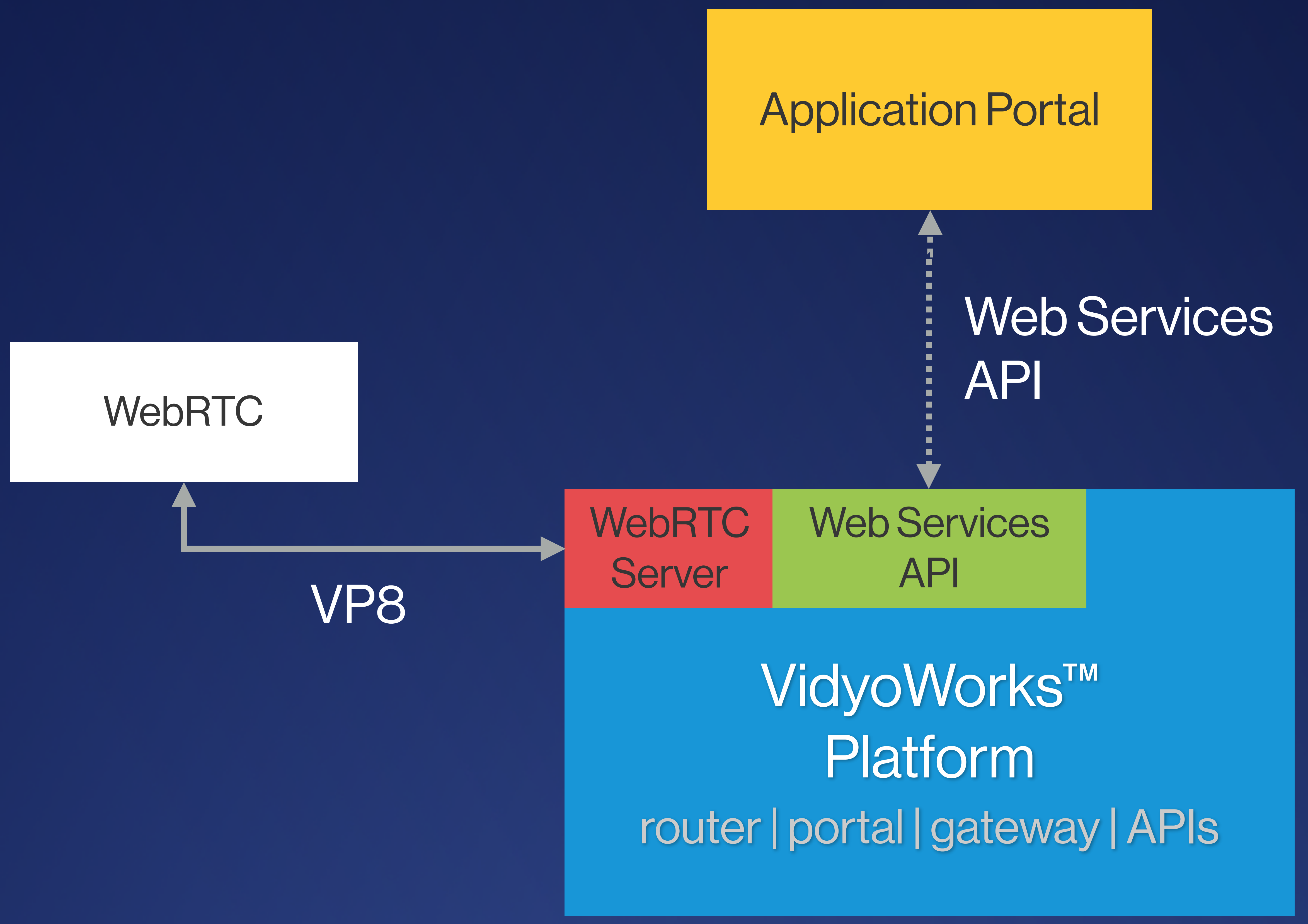
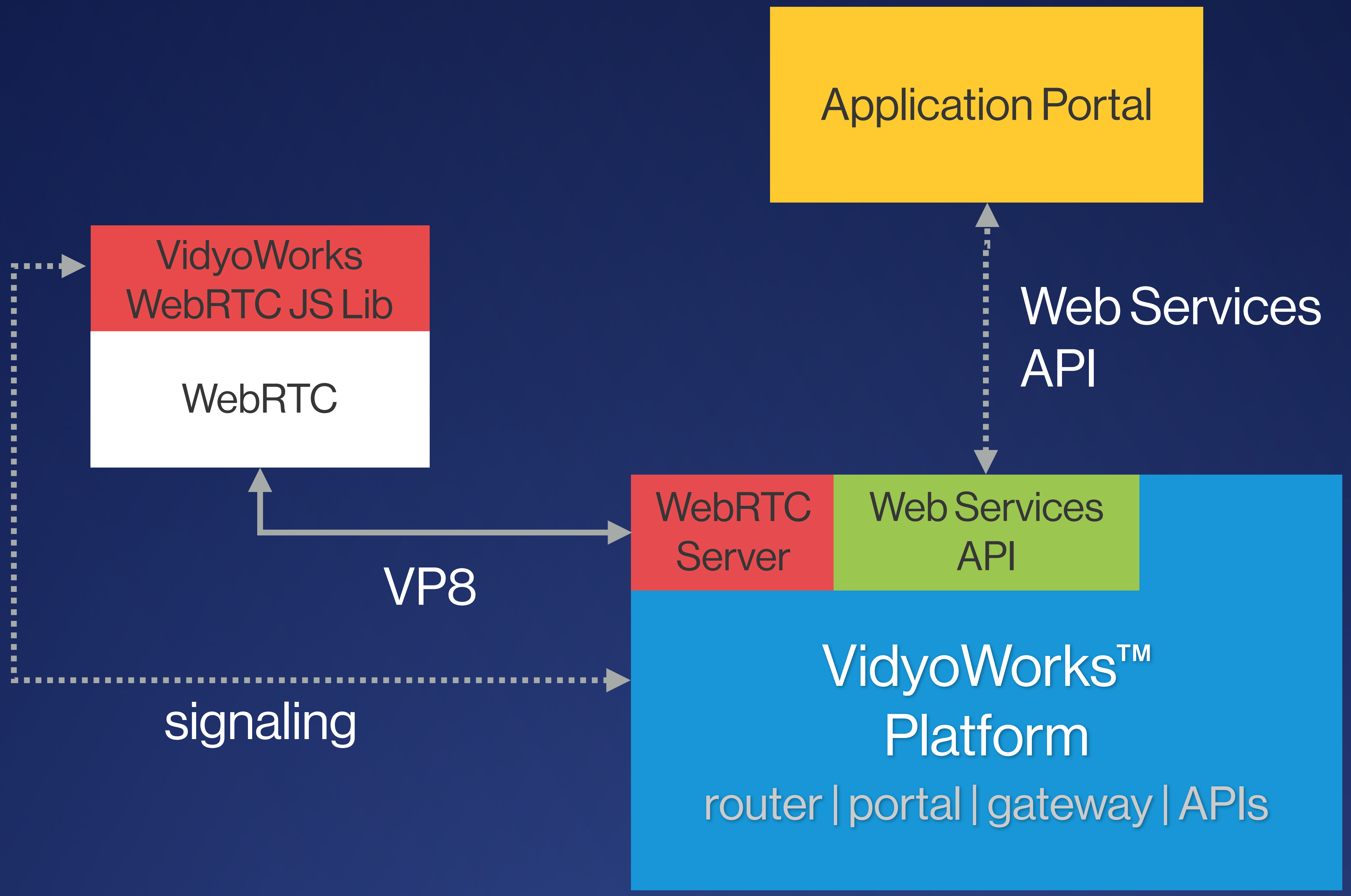

Application Portal
Web Services
API
WebRTC
VP8
WebRTC
Server
Web Services
API
VidyoWorks™
Platform
router | portal | gateway | APIs



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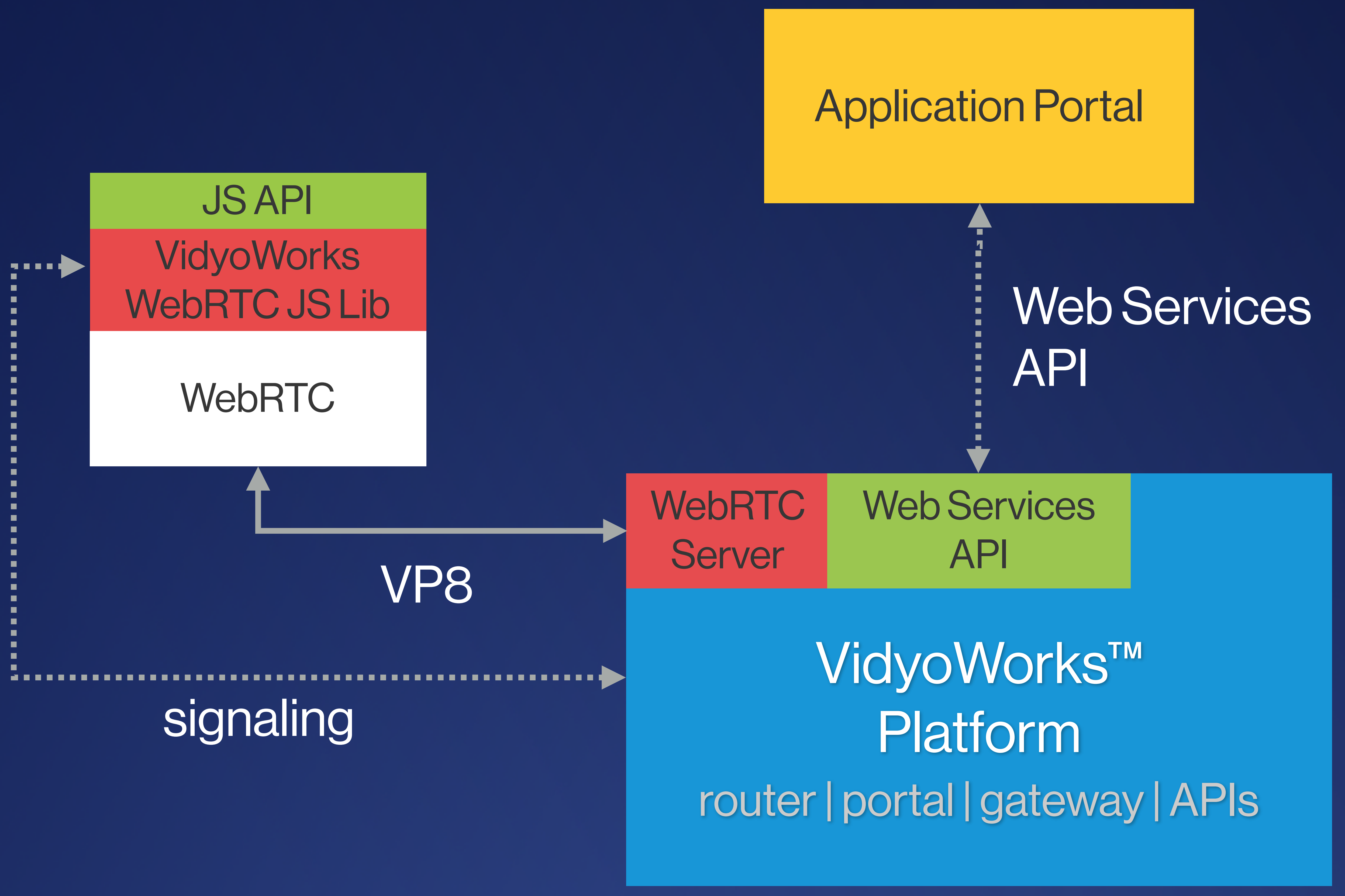
Application Portal
JS API
VidyoWorks
WebRTC JS Lib
WebRTC
Web Services
API
WebRTC
Server
Web Services
API
VP8
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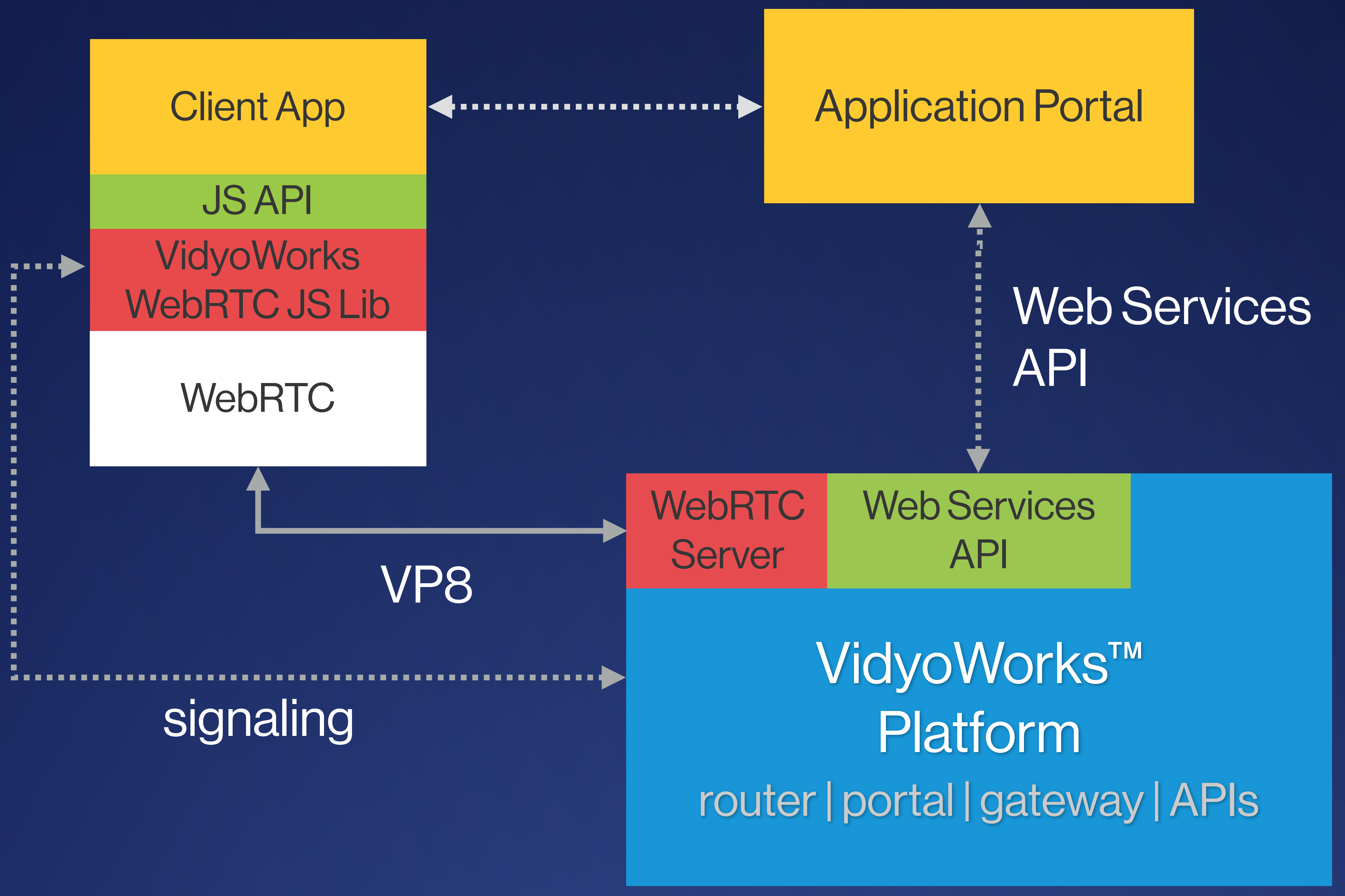
Platform
router | portal | gateway | APIs
signaling


Client App
JS API
VidyoWorks
WebRTC JS Lib
WebRTC
Application Portal
Web Services
API
WebRTC
Server
Web Services
API
VP8
VidyoWorks™
Platform
router | portal | gateway | APIs
signaling

Version 3.0 with sample API released Nov. 2014

Version 3.2 with Vidyo-branded client
to be released May 2015

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alex@vidyo.com

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