To Attend:
• Tutorial 1 day only - $150
• 3 days with Tutorial - $400
• 2 days without Tutorial - $300
Discount Tutorial: $100
Conference & Tutorial Package total: $300

From SIP VoIP to Real Time Web Communications and Applications
October 4 from 8-5
Taught by Alan Johnston and Henry Sinnreich

SIP VoIP is the established world standard for VoIP in the telecom and IT industries alike. Recently however the Web is extending its reach from countless applications and platforms to communications as well. This is also reflected in emerging work in standards bodies, such as the IETF and the W3C. Web real time communications (RTC-Web) enable the seamless match of applications and communications, thus unleashing the potential of creativity of tens of millions of developers for the benefits to users globally.

In this full day tutorial we review the basics of Internet SIP VoIP, the basics of Internet applications, the Web and paths of their convergence. Examples and demos of real-time Web communications are also provided, together with an outline of options for design for both fixed and mobile users.

The tutorial is targeted to a wide audience of engineers, managers and decision makers interested in the convergence of fixed and mobile Web applications and communications.

The authors are well known co-authors of several key Internet standards on SIP VoIP and have a vast experience both in the telecom, web application and tools industries from which they draw the material for this tutorial.

Dr. Henry Sinnreich has worked most of his career in the telecom industry, including 24 years at MCI where he was an MCI Fellow. He has contributed to the development of the first commercial SIP service by a major carrier and is also an active contributor to IETF SIP standards work. He is an author of several books on SIP. Dr. Sinnreich is a guest lecturer at Southern Methodist University in Dallas and works at SNOM Technology AG.

Dr. Alan Johnston has worked in various positions in the telecom industry and is a major contributor to various SIP standards in the IETF that include the core RFC 3261 specification. He has authored several RFCs on SIP service examples that are used as reference throughout the industry and is an author or co-author of four books on SIP and security. Dr. Johnston is an adjunct at Washington University in Saint Louis and works at Avaya on SIP standards and their applications.

www.cpd.iit.edu/voipconference

To register, please visit
www.cpd.iit.edu/voipconference or contact Carol Davids at davids@iit.edu