Utilization with EMS & UPD

The software packages utilized to manage the Code Blue InterAct 3000/3100 phones are Unit Programming and Diagnostics (UPD) and Event Management System (EMS). These software packages were developed for utilization on Key/PBX systems as well as standard P.O.T.S. lines commonly called 1MB (measured business) lines. During the design of the phones and the software packages the ITU-T DTMF standards were utilized not only for the programming of the phone, but also for identification purposes and communication between the phone and the software packages. The timing, duration, and accuracy of the DTMF tones are critical to the proper operation of the software packages.

The Internet and other Voice over Internet Protocol (VoIP) packet switched networks are increasingly used as a transmission medium for voice telephone calls. In order to facilitate communication over these systems, various industry and international standards bodies have established different functional requirements and rules that govern transmission of data packets. Implementation of these common rules, known as “protocols,” is necessary to allow equipment provided by different manufacturers to inter-operate.

One typical device within a packet network is a gateway (also known as an Analog Terminal Adapter or ATA). Gateways allow dissimilar computer networks or telephony systems that might use different protocols to connect with one another. One type of gateway is an Internet Protocol (IP) telephony gateway. A typical IP gateway designed to handle telephone calls can handle multiple simultaneous calls from standard telephone connections originating within the Public Switched Telephone Network (PSTN), and route them over packet networks.

In a typical VoIP connection, a caller places a telephone call using a standard telephone. The call is then routed to a local “originating” telephony gateway which is connected to the VoIP system. The originating gateway then establishes one or more IP “sessions” with a remote or “terminating” gateway that services the telephone at the other end of the call. The terminating gateway then completes the circuit by connecting to the destination telephone via a local circuit switched network connection.

In order to communicate voice audio signals in an Internet-based telephone system, the gateways operate on audio signals received from and transmitted to the parties’ telephones. These audio signals are typically digital Pulse Code Modulated (PCM) signals that may be formulated according to various standards. At the origin point, equipment is used to sample digitize and encode an analog voice signal. The encoded bits are then arranged into packets for transmission over the packet networks that provide the virtual connection. At the termination point, other equipment dissembles the packets, decodes the sample bits, and converts them back to an analog voice signal again. One transport protocol often used for carrying VoIP voice packets between gateways is the Internet Engineering Task Force (IETF) Real Time Transport Protocol (RTP), as defined in Request for Comment (RFC) 1889. RFC 1889 has now been placed into the International Telecommunications Union’s (ITU) standard H.225.0.

There has been a challenge; however, in determining how best to carry push button tones, technically referred to as Dual Tone, Multi-Frequency (DTMF) digits. DTMF digits are typically generated as sequences of sine waves, either added or modulated on the voice signal. DTMF digits are now almost universally used as dialed number digits to establish a telephone call connection. Correct transmission of DTMF digits is also important to ensure the proper operation of the Code Blue units and software packages.

In a VoIP connection that uses the RTP protocol, each end of an RTP trunk typically encodes the voice samples with an appropriate coding scheme, such as the so-called G.711, G.723 or G.729 codec’s. Equipment that uses linear codec’s such as G.711 do not pose a problem since they can faithfully pass the DTMF tones end to end. However, non-linear codec’s such as G.723 and G.729 introduce compression to the digital samples. Therefore, these codec’s do not pass DTMF tones reliably.

Thus, a gateway presently has two options for handling DTMF digits. First, it can use only a linear codec and make no attempt to handle DTMF tones differently from voice samples. However, when compression codec’s are desirable, an originating gateway can detect and recognize an individual DTMF digit and translate it into a data value. The data value can then be encoded into a special type of packet. Upon receipt of this packet at the terminating point, the receiver can then reproduce the corresponding DTMF tone signal. For example, DTMF tones and other named telephony events (such as modem tones, fax tones, etc.) can be generated packets containing as data values, as described in the proposed standard known as RFC 2833. RFC 2833 is an IETF “standards track” proposal for carrying DTMF digits as RTP packets. According to this standard, the packet includes a data value indicating the particular DTMF digit, as well as a volume and duration for each DTMF digit.

Time duration of from 60 through 80 milliseconds (ms) has historically been considered to be the minimum sufficient for reliable transport of DTMF tones through a circuit switched network. However, a common expectation is now that originating equipment can transmit a minimum DTMF tone duration of as little as 50 ms, and that receiving equipment should be capable of detecting any DTMF tone of at least 45 ms. Such requirements are, for example, promulgated by the International Telecommunications Union (ITU) in the Q.24 specification. An additional difficulty is presented by the fact that it is not always possible to playback tones at the destination with the same duration originally created at the origin point. Thus, even if the tone duration can be correctly reproduced, other impairments in the network introduce an additional complication. Such impairments are typical of IP networks even though QoS, Vlans, protocols, latency buffers and other methods have been implemented to ensure voice packet delivery and call quality.

The effects of the network on EMS and UPD functionality can be summarized as dependant on the capabilities of the network. One area of focus would be the architecture of the network. The implementation of Vlans, QoS, and other aspects such as bandwidth control on trunk ports will critically impact the timing of the DTMF signaling. Network congestion at certain times of the day, during network outages, or application requirements may cause failures due to EMS or UPD not receiving the DTMF signals in the required timeframe. Gateway configuration is another area to pay specific attention to as well. Some manufacturers provide the ability to control many functions of the analog ports. Key points to keep in mind are that Code Blue units and software utilize FXS ports and also “listen” to the signals on the line to identify dial tone, DTMF signals, call progress tones, and wink signals. The acronyms and definitions of these terms may vary from the traditional telecommunication system meaning to the VoIP system definition. Wink commands are a temporary reversal in polarity, typically 200 ms. Call progress tones are also call ring back tones and is the on-off ringing you typically hear after dialing a phone number. There are other tones such as re-order tones that repeat when the phone is left off hook to long without dialing. Some VoIP systems will term this as warning signal/tone etc.

New College of Florida recently installed the Code Blue UPD software to manage the Code Blue InterAct (IA) 3100 phones installed on the campus. The VoIP system and network equipment were all 3Com (VCX, 8800, 4500). Network engineering and configuration was optimized for the VoIP traffic during the design phase. During the installation of the UPD and InterAct 3100’s several detailed conversations ensured concerning the configuration of the V7111 gateways used to connect the UPD and IA3100 to the VoIP system. Once the correct settings were determined the functionality of the units and software was flawless.

Many other installations, including Cisco Callmanager and Avaya Open Office, of the InterAct 3100 and UPD/EMS software packages have been successfully installed on VoIP systems. The key is to understand the DTMF timing issues as well as other signaling types that are typical to the traditional telephony network and apply that knowledge to the configuration of the gateways. Engineering of the network infrastructure to ensure QoS from end to end is essential for the timing as well.

Code Blue is ready to help in understanding and configuring your particular installation. As with all VoIP installations where the EMS/UPD software is also utilized, Code Blue Corporation cannot guarantee the network or VoIP systems operation so it is suggested that a trial of the software be obtained before actual purchase.