Application Note

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Voice over IP Testing with Domino® Internetworking Analyzers

H.323 is a family of specifications that define real-time multimedia information transport over IP networks. This application note provides information on using Domino internetworking analyzers to identify problems with signaling and call quality on Voice over IP (VoIP) networks that are H.323 compliant.

H.323 VoIP Components

The H.323 specification defines the components and protocols that combine to make up a VoIP network. Below is a list of H.323 components, the protocols and their functions.

- **VoIP Terminal**
  Any device that supports the H.323 specification to place and answer calls can be considered an H.323 terminal. This typically refers to Ethernet IP telephones or PCs with software and hardware that supports VoIP capability.

- **VoIP Gateway**
  A gateway's job is to connect different types of networks. Gateways are generally used to allow VoIP calls on a LAN to be connected to a different network, such as the PSTN or a private Voice over Frame Relay network. To do this the gateway must translate the signaling and codec type between the two different networks.

- **Multipoint Control Unit**
  The MCU is a device used for conferencing among multiple terminals. During a conference call all of the terminals involved connect to the MCU. Call parameters such as codec type are controlled by the MCU.

- **VoIP Gatekeeper**
  The VoIP gatekeeper controls a VoIP zone. A VoIP zone consists of a number of gateways, terminals and MCUs. Along with supporting the RAS functions described below the gatekeeper may also perform billing and other management functions.

  The terminal, gateway, gatekeeper and MCU are all logical device designations. A single physical device can perform more than one of these functions.

H.323 VoIP Protocols

- **H.225 Call Signaling Protocol**
  Borrowing from ISDN, the H.225 signaling protocol is based on ITU Q.931. Anyone familiar with ISDN D channel signaling will notice similarities between Q.931 and H.225. The purpose of signaling protocols is to define the messages sent between end stations to establish and tear down a connection. This includes information related to the calling party the called party phone numbers, call type (voice, data, etc.) and the encoding method used for the payload information.

- **H.225 RAS Protocol**
  In addition to call signaling H.225 is responsible for Registration, Administration and Status messages. Before a call is set up the endpoints use RAS to exchange information with the gatekeeper for administrative purposes.

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- **H.245 Control Signaling Protocol**
  VoIP endpoints use H.245 to perform end-to-end control-specific tasks. These relate to functions such as capability exchange and opening and closing media stream channels, among others.

- **Real Time Transport Protocol (RTP)**
  RTP carries the digitally encoded voice information between VoIP end stations. RTP is defined in IETF RFC 1889 and ITU specification H.225. The actual digitized voice payload is encoded using one of the ITU G-series codecs described below.

- **Real Time Control Protocol (RTCP)**
  RTCP is a series of control messages exchanged between the VoIP end stations during a call. RTCP messages contain information related to the status and quality of the connection. The RTCP protocol is defined in IETF RFC 1889 and ITU specification H.225.

H.323 Audio Codecs

A codec (COder / DECoder) is a device that converts analog voice information into digital information.

- **G.711 PCM Codec**
  ITU G.711 is the Pulse Code Modulation (PCM) 64 Kbits/s voice encoding method currently used by telephone companies around the world. Although it is included within the H.323 specification, it is the most bandwidth-hungry method of encoding voice and is therefore not often used with VoIP systems.

- **G.723.1 Low Bit Rate Codec**
  Based on more advanced principles of predictive audio encoding, the G.723.1 codec can provide acceptable voice quality at data rates of 5.3 Kbits/s or 6.3 Kbits/s. Although it offers very low bit rates it requires more complex algorithms, and thus greater encoding and decoding delay, than most other encoding schemes.

- **G.729 Low Bit Rate Codec**
  Using principles similar to G.723.1, this codec is less complex and offers data rates of 8 Kbits/s. The result is a much reduced data rate when compared to G.711, with less processing delay than G.723.1.

  Other less commonly used codecs are also supported by H.323.

Other Relevant standards

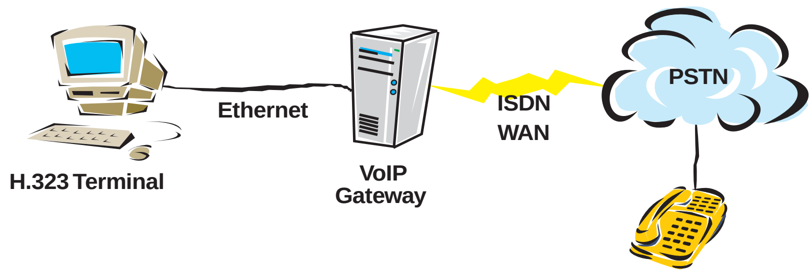
Along with the standards listed above there are a number of other standards within H.323, as well as alternatives to H.323. These include alternative signaling methods as well as data sharing, fax, video conferencing and security standards.

Identifying Signaling Problems

In order to allow communications between the LAN and PSTN, both the signaling and the audio encoding may need to be translated. Identifying the source of signaling and audio problems can present a challenge.

The H.323 user terminal places a call by sending a Setup message to the gateway. It is the gateway's responsibility to translate that call into the appropriate signaling for the PSTN. As shown in Figure 1, the PSTN connection is ISDN so the signaling would be a country specific variant of Q.931.

The Setup message contains information elements that define the characteristics of the call. These characteristics include the called party phone number, whether the call is intended to be speech or data, and any additional information required.



Several possible problems can cause a call attempt to fail. Failure of the gateway to properly translate the Setup message elements between the VoIP terminal and the PSTN network may be at fault. Problems can also stem from the interaction of the ISDN network and the gateway configuration. Another problem could be the inability of the called device at the receiving end of the call to accommodate the requested call parameters defined in the setup message. The end result for any of these problems is a failed call. Identifying the source of the problem can be a challenge.

The signaling method used with H.323 is called H.225. This ITU signaling specification is derived from the Q.931 standard. Because of this relationship there are fields within H.225 that look very much, if not exactly, like ISDN signaling. The problem is that with H.323, much like Q.931, vendor implementations are sometimes incompatible.

The most efficient setup for identifying signaling translation through a VoIP gateway is shown in Figure 2. In this configuration two Domino analyzers are daisy-chained to a single PC. While any appropriate member of the Domino family of analyzers can be used to monitor the LAN and WAN segments, in Figure 2 the DominoLAN and DominoWAN ISDN are shown.

In the example, the DominoWAN ISDN is configured to monitor signaling on the D channel. The DominoLAN has a filter set to capture only H.245 signaling packets. Configuring the analyzers in this fashion eliminates all traffic not involved in call signaling, making the side-by-side comparison of the two capture files easier.

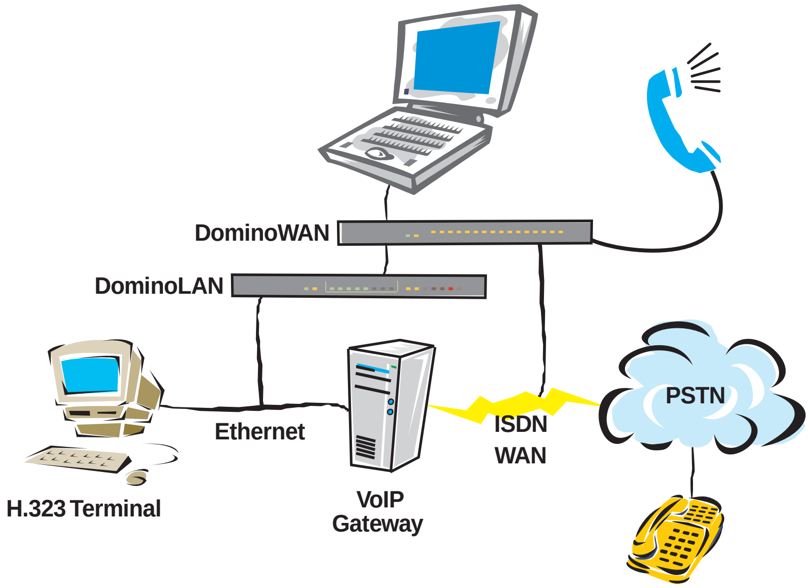


Figure 2: *Dominos set up for synchronized data capture on both the LAN and WAN sides of a VoIP gateway.*

With the Dominos capturing signaling frames, a phone call can be attempted between the H.323 terminal and a telephone attached to the PSTN. Once the H.323 terminal has reported the results of the attempted connection, use the Examine feature on both Dominos to match the VoIP signaling protocol on the LAN side of the gateway to the Q.931 country variant on the ISDN side of the gateway.

When multiple Dominos are stacked as shown in Figure 2, the Dominos operate with synchronized timestamps. A maximum of eight Dominos can be stacked in this way. The synchronization ensures that, when displayed, the user can easily identify related frames. An additional advantage is the ability to determine the latency imposed by the gateway on the translation and processing of the signaling frames.

Figure 3 shows an example of a dual display of the H.323 setup message and the ISDN setup message. The top half of the display is the VoIP setup message. The bottom of the screen shows the ISDN setup message.

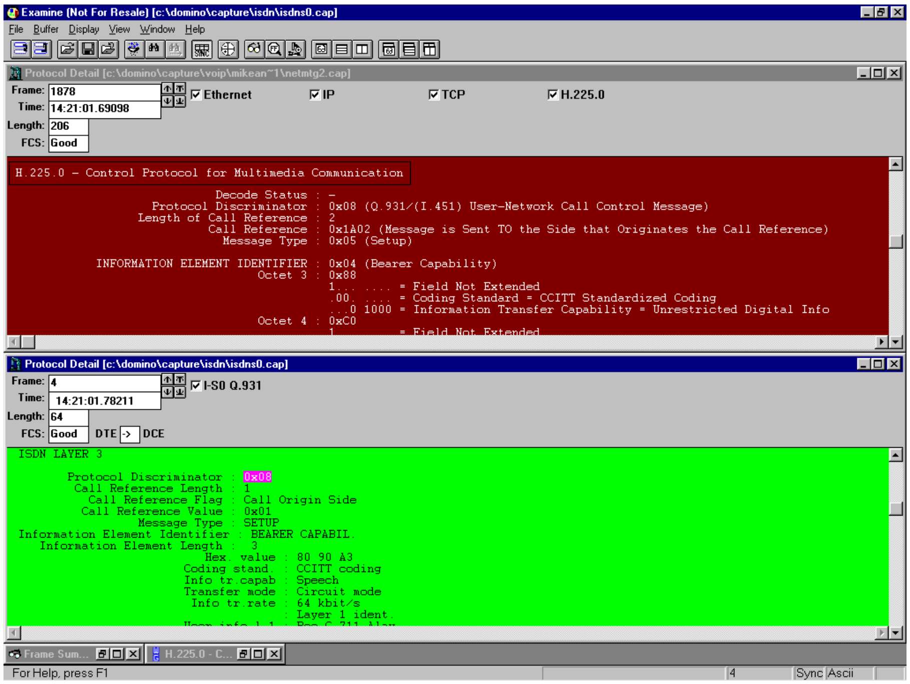


Figure 3: *VoIP Setup Message (top) and ISDN Setup Message (bottom)*

The messages needed to set up a call in H.323 are very similar to those in ISDN. Figure 4 shows the H.323 and ISDN signaling message flow. The RAS protocol and H.245 connection control protocol have been omitted for simplicity. It is important to ensure that all of these processes operate properly in order to setup a connection.

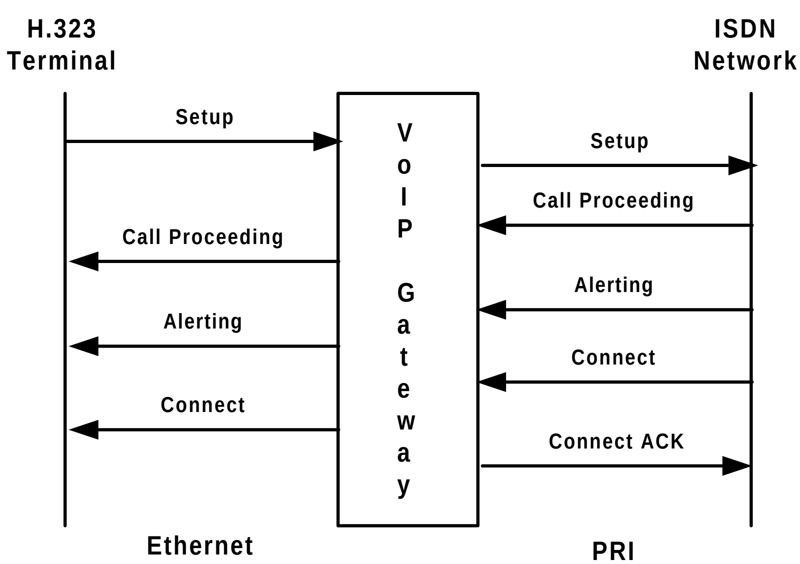


Figure 4: *H.323 and ISDN Signaling Flow*

Monitoring Audio Quality

As VoIP networks become more common so do the problems associated with interconnecting to the Public Switched Telephone Network (PSTN). Combining aspects of voice and data communications problem isolation and resolution can present a challenge. A basic understanding of the responsibilities of the components in a VoIP system can help in tracking down the source of the problem.

When calls are placed between the VoIP network and the PSTN, the gateway is responsible for translating the signaling and audio encoding between the two networks. Unlike the traditional G.711 encoding used in PSTNs, VoIP encoding techniques cannot be measured with traditional test sets. With VoIP the concept of connection quality becomes much more subjective, particularly the quality of a voice connection.

The low bit rate codecs that are used with VoIP place more speech information into fewer bits. As a result, packet loss becomes a more significant issue. Some packet loss is to be expected and should not cause intelligibility problems. However, if too many packets are lost speech quality becomes unacceptable.

The DominoWAN Internetwork Analyzer's ISDN interfaces provide a method of judging the quality of a voice connection. They have a built-in G.711 codec for call monitoring purposes. By attaching a handset to a connector located on the Domino ISDN interface module and selecting the B channel to monitor, it's possible to listen to a conversation in real time.

Identifying Voice over IP Quality Problems

To troubleshoot a VoIP quality problem it's useful to locate the source of the problem. If the poor quality signals are coming from the PSTN, placing a number of calls to different locations helps determine if the problem is related to a single location or is occurring at all locations. This method determines if the problem is related to a Local Exchange Carrier (LEC) or the InterExchange Carrier (IXC). If poor speech quality is experienced when the person using the H.323 terminal is speaking, then the problem is probably related to the LAN segment.

Three primary measures are used to identify problems with VoIP transport: delay, packet loss and jitter.

- **Delay**
  Delay is the time it takes for information, in this case speech, to traverse the network from end to end. When the delay is excessive it becomes difficult to conduct a normal conversation. This used to be a common occurrence when placing transcontinental telephone calls. The delay makes each party in the conversation unsure if the other party is simply pausing or if they are waiting for a reply. As a result both parties begin to talk at the same time.

  Overall delay in VoIP networks is made up of a number of smaller delay-inducing tasks that must be performed between the speaker and listener. These tasks include converting analog information to digital format, placing the digital data in packets, transport delay (routing, switching, transport), processing of received packets, jitter buffer-induced delay, and converting digital information back to analog (audio).

Overall delay should be kept as low as possible. In general, a delay of less than 200 milliseconds (ms) is required for acceptable performance. Intermittent increases between 200 and 800 ms can be tolerated for short periods of time.

- **Packet Loss**
  Packet loss occurs when packets generated by an H.323 terminal fail to arrive at their destination. Unlike some other IP-based protocols, RTP has no retransmission facility. The VoIP encoding methods tolerate some loss, but if it becomes excessive voice quality suffers.

  The effects of packet loss are based on the number of milliseconds of speech interruption that the codec can tolerate. RTP packets have a fixed size, and contain a defined number of milliseconds of speech. As a rule, about 20 milliseconds of speech loss can be tolerated for short periods; the specific level of packet loss tolerance depends on the codec in use.

- **Jitter**
  Jitter is the variable delay that RTP packets experience when traversing the network. During a call individual RTP packets experience different amounts of delay due to varying network conditions. To accommodate this delay variation, the H.323 terminal buffers incoming data. It then plays the data from the buffer in real time, eliminating the delay between individual packets that was experienced during transport. However, the jitter buffers add a small amount of delay to the overall signal transit time.

  Jitter in VoIP networks results primarily from variations in the network transport. Because a VoIP network is typically LAN-based it has shared segments, shared components, or both. Changes in overall network traffic volume will affect the transport speed of each packet.

The Domino analyzer is able to identify the delay, loss and jitter being experienced by each H.323 terminal by decoding Real Time Control Protocol (RTCP) packets. During a VoIP call the terminals involved in the call use RTCP packets to exchange status information. By decoding the RTCP packets, the Domino analyzer can identify circuit-related problems being experienced by an H.323 terminal.

In the setup shown in Figure 2, RTCP sender reports are exchanged between the H.323 terminal and the gateway. The ISDN side of the gateway is operating in accordance with the rules of the PSTN, not H.323. When the Domino analyzer is positioned as shown in Figure 2, it can capture the RTCP packets on the LAN segment. By decoding the RTCP packets the jitter, packet loss and delay being experienced by the gateway and H.323 terminal can be seen.

RTCP sender reports are transmitted periodically during a VoIP call. The frequency of the sender reports varies with network conditions, as opposed to being sent at fixed intervals. In general, each H.323 terminal participating in a call generates RTCP packets every 3 and 10 seconds.

Several fields are of interest in the RTCP packet, shown in Figure 5. The sender report portion of the packet contains information about the number of packets and bytes of encoded voice data that an H.323 terminal has sent. Delay information can be determined if the H.323 terminals are using a common timestamp through NTP. If not, the timestamp information is of questionable value and should not be viewed as an absolute indicator for use in delay measurements. The reception report portion of the RTCP packet contains information about the jitter and packet loss associated with the received data stream. This can be valuable in identifying quality problems related to packet transport issues.

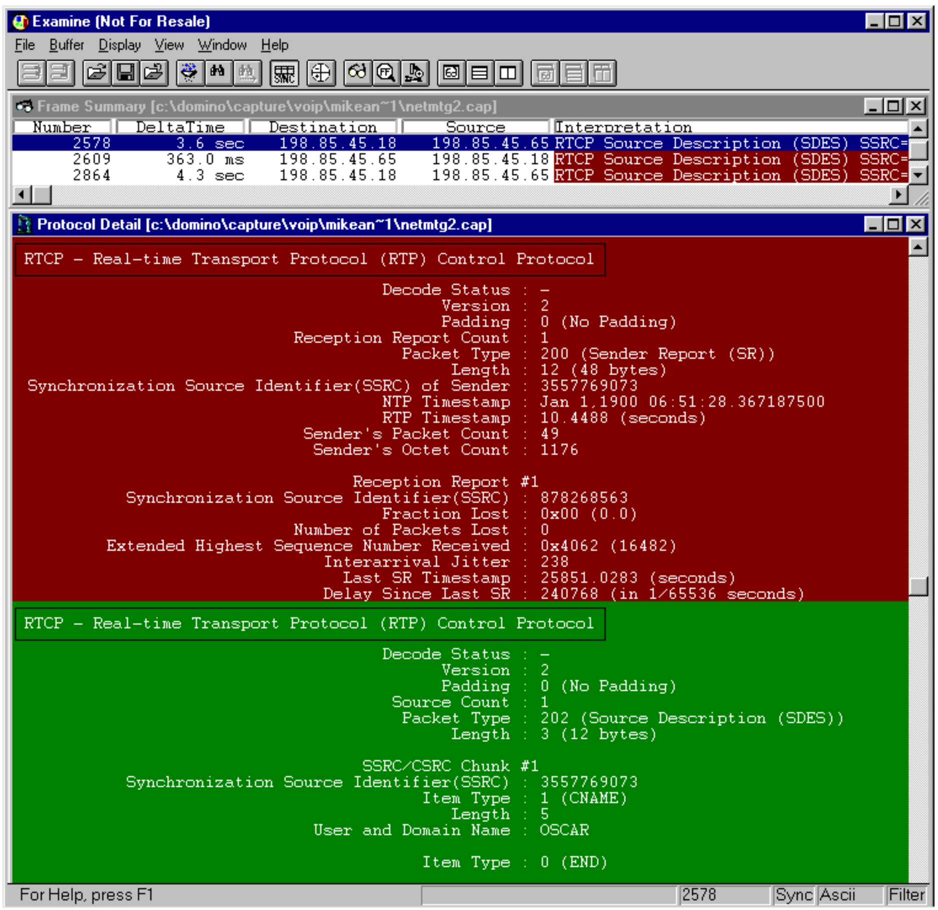


Figure 5: *Detail of an RTCP Packet*

Capacity Planning for VoIP

A baselining tool can help identify whether the overall network traffic load is having an adverse affect on the VoIP traffic. WWG's Wizard baselining application can be used with the Domino analyzers to evaluate the network's overall utilization pattern (Figure 6).

Baselining the network can be useful before VoIP is deployed on the network. Baselining can identify segments that may exhibit problems transporting real-time VoIP traffic in advance actual deployment. To ensure problem-free operation with VoIP, the network may need to be segmented or Ethernet switches may need to be added.

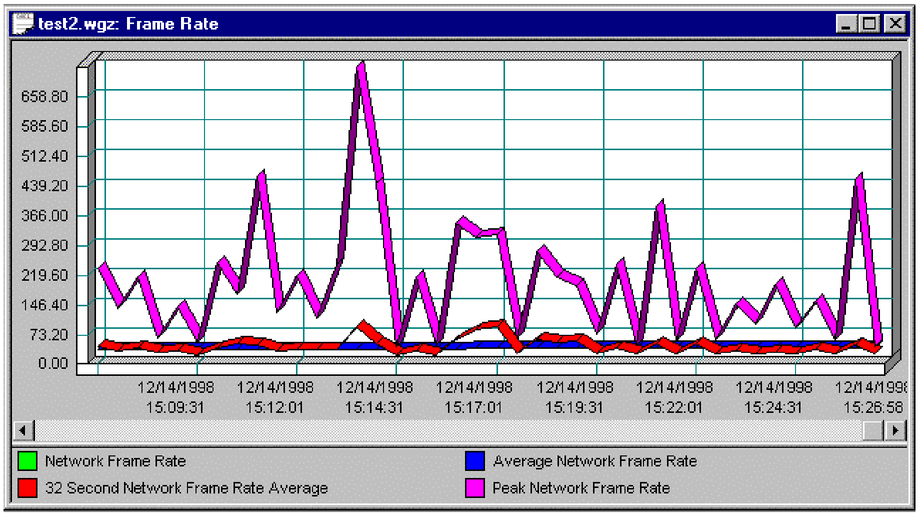


Figure 6: *Example of WWG Wizard Graph*

Domino Family Voice Capabilities

Typical network analyzers have the ability to perform data traffic decoding but cannot perform analysis of encoded audio information. Traditional transmission impairment test sets (TIMS) can handle end-to-end testing of PCM encoded audio but lack useful detail of signaling decodes. In order to address this issue the user needs a product that is at home in both the telephony and data environments.

While a variety of analyzers support frame capturing and decoding, few are adequately equipped with the necessary features to perform the task in a truly integrated and efficient manner. Domino Internetworking Analyzers can be a valuable aid in capturing and decoding the signaling on the LAN side of a gateway as well as the PSTN, ATM, Frame Relay or WAN side of a gateway simultaneously. By connecting two Dominos to a single PC users can track frames through a VoIP gateway.