

# Anomalies in H.320 Videoconferencing

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Even during the "rush" to IP Packet-based H.323 communications, many people are continuing to run Circuit-Switched H.320 solutions, in-whole or in-part, as they learn about and begin to migrate to an H.323 or SIP-based enterprise, and will need to run some level of H.320 for some time to come.

This means that end users must continue to work and "cope" with certain elements of the global Circuit-Switched digital network that are often inadequately documented or downright confusing.

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# Anomalies in H.320 Videoconferencing, Section 1

## Introduction and Background

This document is designed to explore certain elements of ISDN-based H.320 communications and assist in determining potential causes of [and solutions for] anomalies that can and often do occur when using modern H.320 capable terminals connected to and communicating through Circuit-Switched Digital/ISDN BRI (Basic Rate Interface) and PRI (Primary Rate Interface) Services.

The origin of this review and discussion comes from work that has been [and is being] done by many people who are still using Circuit-Switched/ISDN connectivity for some or all for their videoconference communication connectivity, using the H.320 protocol suite.

### Issues and Problems

In the past year or so I [Scott Sharer] have been able to ascertain, handling actual troubleshooting calls and in conversations with a large number of people, that there is an increasing perception of certain performance deficiencies [problems] when using an H.320 videoconference terminal over ISDN connections. The issues and problems raised are not "new" (I have encountered them two or three times a year for over 15 years now), but the incidence rate for a specific small set of these issues has dramatically increased in recent months (up to 73 in the past 10-14 months to be exact). Though I am not completely certain why the dramatic increase has occurred, I do know that there is vast confusion and misunderstanding about the nature and origin of the problems, and, as a result, difficulty on the part of even very capable people as they work to resolve these problems. Specifically, the issues and problems stated to me are most succinctly summed-up as follows:

1. "Certain of our H.320 terminals within our enterprise have a tendency to engage in Downspeed activities even when there are no apparent network anomalies that might induce such activities." ... and/or ...
2. "When using our codecs/terminals for videoconference communication over ISDN Circuit Switched connections, there appears to be some degradation in the visual video information as displayed on the local terminal after receipt and decode from the unit or units connected at the far end(s) of the call". Specifically, it has been stated that there has been "observable tiling, "jerkiness" and inadequate picture resolution in the received decoded video, appearing at random times and in random manners, and connections must be terminated and re-established multiple times during a single meeting to remove the anomalies."

**While difficult to definitively state what is happening in any specific enterprise, it is my professional opinion that, for many people expressing these complaints, there may be more than one problem or causative element at-play.** I will attempt to pull-apart the underlying elements and issues related to these two (2) stated performance deficiencies, explain [as clearly as possible in a document of this type] the possible nature of the origin of these deficiencies, provide some processes or steps that can be taken to validate or nullify our diagnosis and [finally] resolve, if validated, the issues in a positive and permanent manner. **Also Note: These two different individually stated problems are often irretrievably linked together.** The underlying causes of one problem may also create the other, and vice-versa. In this instance, I believe that though often different in appearance and effect, they are, indeed, tied together, and will proceed with my writing holding this premise.

**Please note: As is our custom at CDG Inc, our only goal is to properly define and then resolve these issues for the end-user community.** Additionally, as we so often find in our work with our large Fortune 50 clients, and their very capable technical groups, the path(s) to our goal will require, for any reader of this document, a little background and education on the nature of the encoding - decoding process, the "undocumented idiosyncrasies" of the Circuit Switched [ISDN] network, and some slightly "off the beaten path" directions in our thinking as it relates to actually troubleshooting these problems. There is a potential for this to be a far-ranging and wide-reaching discussion, and, as such, this could easily become a convoluted and unproductive mix of topics. I will attempt to keep the discussion in-focus and on-track, avoiding the random and ancillary topic paths that too often lead to more confusion and misdirection, not less.

**For anyone or any group experiencing these problems,** and for those who have spent countless hours and [possibly] dollars trying to resolve them: **I am not writing this document for the purpose of proving any past theories or disproving any past discussions or solutions that have been presented to you.** I am only interested in discussing this set of anomalies within the context of what we at CDG Inc., as a professional design and

engineering firm, have encountered over the last 20 years out there in the field while working with the same types of devices and services.

**Some of the information we are about to present and discuss** will be familiar. Certain specific portions of the materials, concepts and issues that we are going to present will appear to be unique, undocumented anywhere in any known reference. That is because some of the elements are, in point of fact, based on those tidbits of ancillary information that are gathered and applied consistently over extended periods of time through hands-on activity. I am speaking here of the type of information that was and is frequently derived through [often unorthodox] thinking patterns that do not always follow strict engineering formulas.

**Here's the deal** - sometimes things don't work the way they are "supposed" to work and, unless you know that this is the case, you may find that you are unable to properly and accurately define the root-cause of a particular problem or problem set. This then results in an inability to find a fundamental solution to overcome that root-cause, even when the solution is being sought by highly educated and capable people.

**As much as possible** we will attempt to be straightforward and direct, providing technical discussion points only as far as necessary in order to properly define or describe any specific point or issue in the search for clarity and resolution, and we will make every effort to "hit-the-mark" for each reader of the document. That being said, to some advanced technical specialists with many years of experience in Circuit Switched Digital for Videoconferencing this document may appear at times to be a bit simplistic. To others who, though very capable, may have less experience in the underlying elements of this environment, this document may be a bit difficult to "wrap your head around". We will, hopefully, strike a median level of the discussion so that nobody is either alternatively bored or confused on the details of the topics.

## **Anomalies in H.320 Videoconferencing, Section 2**

**We need to begin by understanding the nature of the "Downspeed" mechanism**, why it is present as a function in modern terminals, why it was created and why it is still necessary, and what the situations or circumstances are that cause a terminal to invoke this mechanism.

**"Downspeeding" is essentially the renegotiation of the connection speed or bandwidth of the call** that occurs between two endpoints or terminals any time there is a complete loss of channel synchronization within one or more channels in the connection, after the initialization and actual start of the call (after the end-user data begins to transit the connection). Downspeeding was pioneered by Tandberg Inc in the early-to-middle 1990's and was quickly adopted by other manufacturers.

**Downspeeding was and is designed to create call stability in the face of unstable network transport conditions.** Prior to this mechanism, if there was a loss of synchronization between the terminals on one or more channels after the start of the call, the entire call would "disconnect" (the terminals would "hang-up" the call).

**The thinking behind Downspeeding was [is] essentially this:** Even if the caller prefers to have the connection for their video meeting run at, for instance, 384kbps (6 Channels at 64kbps per channel), in the face of the loss of one or more of those channels it would be better for the end-user (the people who are attempting to have an uninterrupted exchange of information) if, instead of dropping or disconnecting the call and forcing the meeting participants to attempt an entirely new connection (dial the call all over again), the call speed could be renegotiated quickly on-the-fly, and the connection could be maintained instead of completely dropped. Though the newly negotiated connection would operate at a lower bandwidth or speed, it was believed (and rightly so) that the end-users would prefer to have their call slowed from, for instance, 384kbps to 256kbps rather than have the call tear-down entirely. This allowed the end-users to maintain better flow in the human conversational aspect of the meetings. It was [and is] correctly believed, by standards of measure, that any resulting changes to the video and/or audio as a result of the slower speed of the call would be less objectionable than losing the connection entirely. [Just ask the users.]

**This was an enormous leap in the perceived durability and stability of switched-digital connections through the digital circuit switched network.** Downspeeding was quickly embraced by the user community [and VTC manufacturing community] as a dramatic improvement over the traditional "we have to redial the call many times during a single meeting because the systems keep hanging up". Even today, with the now mature digital circuit switched network facilities present around the world, there are huge pockets of this network that remain "brittle" and unstable [particularly in the developing world and even in developed areas of the world where rapid population increases and congestion have overtaken network capacity], and Downspeeding continues to provide the type of stable connections necessary to satisfy human communications, even in the face of often serious network instabilities.

**Now that we have established what this mechanism is and how it is helpful, we can begin to discuss the underlying reasons behind why it gets invoked.** This is important to us since it begins to get to the heart of some of the problem-issues we confront in a Circuit-Switched H.320-based videoconference system or enterprise.

We have stated that this mechanism was created to handle network problems that were inherent in the original digital ISDN deployment [and still handles some of the problems that remain today], especially in timing and synchronization. Loss of synchronization can happen as a result of a number of different difficulties within the network itself, and also within the deployed enterprise we are managing.

**Generally, when we see an unstable ISDN-based network or network segment (BRI or PRI),** the instability is actually a result of extreme jitter (the variance of the "clock" from which network and data timing is derived) in the carrier. **"SearchNetworking.com" defines Jitter as:** "1. ...the variation in the time between packets arriving at a destination system or device, frequently caused [in an IP network] by network congestion, timing drift, or route changes." ...and... "2. In any network system using high-speed digital carrier signals, Jitter is the deviation in or displacement of some aspect of the pulses in a high-frequency digital signal. As the name suggests, Jitter can be thought of as shaky pulses. The deviation can be in terms of amplitude, phase timing, or the width of the signal pulse. Another definition is that it is "the period frequency displacement of the signal from its ideal location." Among the [many possible] causes of jitter are electromagnetic interference (EMI) and crosstalk with other signals [such-as NEXT\* in cable terminations]. Jitter can cause a display monitor to flicker; affect the ability of the processor in a personal computer [or codec] to perform as intended; introduce clicks or other undesired effects in audio signals, and loss of transmitted data between network [or terminal] devices. The amount of jitter "allowed" depends greatly on the application [and the capability or tolerance of intermediate and endpoint devices.]"

\*Near-End Cross-Talk

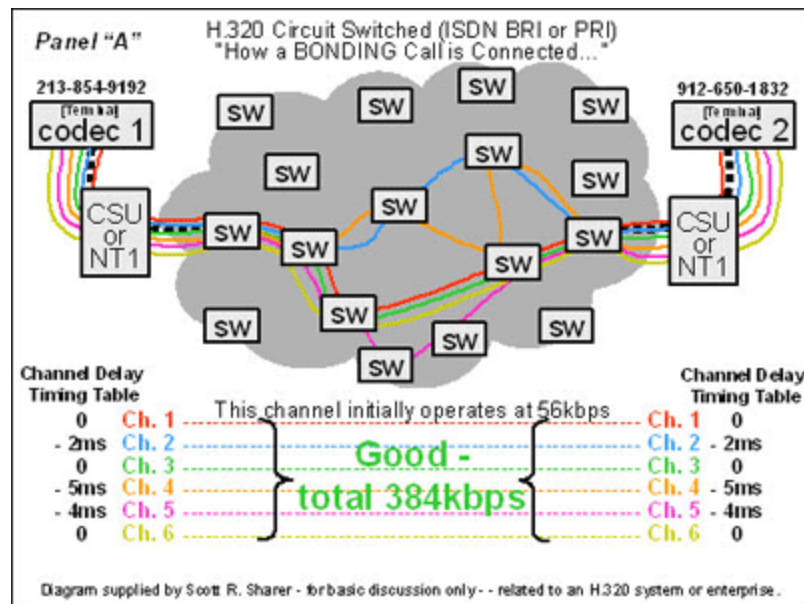
note: "[...]" = comment added by S. Sharer

**Such "jitter" or clock-instability can result in** improper packet alignment or even, in extreme cases, latency in a packet stream that is so far outside the window of acceptability for a given terminal that the terminal is forced to discard the packet or series of packets presented within the extraordinary timing range for the duration of that Jitter-instance. **The result of "jitter"** in an H.320 VTC call can mean loss of terminal synchronization. As noted - jitter can be caused by any number of things.

For our purposes we are interested in only a couple of the possible causes, most importantly a specific cause that can be created by the terminal itself (actually - by us, the human beings, who physically set up and configure the terminal).

## Anomalies in H.320 Videoconferencing, Section 3

To understand some of the critical nature of timing and clocking delay for data in an ISDN Circuit Switched H.320 call, we need to first understand the basic process of call setup and maintenance that occurs during a connection. Only after we understand this process can we explore one or more of the potential causes of mistiming, jitter or improper synchronization that, in turn, tend to be the root causes of user-perceived anomalous manifestations in the decoded and displayed video and audio portion of a videoconference call. For comprehension and illustration purposes, please read the following section(s) while reviewing the included image-panels labeled [variously and in-sequence] "Panel A", "Panel B" and "Panel C" (this label appears in the upper left hand portion of the included graphics). These simple illustrations start with a properly connected H.320 Circuit-Switched ISDN call, then show the effects of configuration or connection inconsistencies on that proper connection, detailing the "effects" that are most often observed on the part of even non-technical end users.



Here we are able to see a completely connected H.320 call through a Circuit Switched/ISDN network. Using the process for call setup known as "BONDING" (Bandwidth ON Demand INteroperability Group - the organization of manufacturers who established this defacto standard prior to a similar standard being issued by the ITU-T), one Terminal "dials" another in-band on one of the Bearer Channels.

**For instance:** In this diagram, let's say that codec #1 dials the number for codec #2 by dialing "1-912-650-1832". The digit "1" means that this is a long distance call, and the call will transit out onto the network, first moving through the local CO [or Central Office] to some portion of the long-distance network and then back to the far-end local network connection, moving out across the circuit switched network and connecting a "path" through a series of switches within the network (for our purposes and for this discussion we are not interested in the various types of switches or the hierarchy of the NAM or North American Multiplex). In our simple diagram here the first or "red" "path" (Ch. 1) gets set-up through five (5) different switches.

**This initial "path"** (noted as "Ch. 1" - the red line in the illustration above) **is the connection for Call Initiation** that "rings" the far-end and across which the far-end "answers". This is also the channel across which the two terminals establish / agree-to the speed of the overall connection and the connection timing (Q.931 Signaling), the directory number(s) to be used and the capabilities of each endpoint to deliver data and decode data (generally referred to as the "CAPS" or "H.245 Capabilities Exchange" - H.264 for video, G.722.1 for audio, etc). **This "setup" information is accomplished across a channel that initially only operates at 56kbps** - even if we are actually establishing a connection of 6-channels at 64kbps each (384kbps).

**Once the first path or channel has established**, the initiating terminal (codec #1) must then place five (5) more connections between itself and codec #2. **If there is a PRI connection to codec #2**, codec #1 will continue to dial the same number that it dialed for the first channel (the PRI serving codec #2 probably has only one DID or Direct Inward Dial number assigned to it), and each channel (Ch. 2 - blue, Ch. 3 - green, Ch. 4 - orange, Ch. 5 - pink, and

Ch. 6 - gold) will be added to the call by codec #2 as it "rings-in". **If there is a set of BRI connections to codec #2**, then codec #1 will dial the individual Directory Numbers (DN) for the Bearer Channels serving codec #2 and, once again, as these channels "ring-in" they will then get synchronized and added into the overall connection or call (Ch. 2 - blue, Ch. 3 - green, Ch. 4 - orange, Ch. 5 - pink, and Ch. 6 - gold). During this process, the first channel also throttles "up" from 56k to 64k. The result will be six (6) channels each operating at 64kbps for a total or aggregate bandwidth of 384kbps.

**If the "connections" subsequent to the first connection followed exactly the same path in the network, then "timing" would be easier to maintain.** In the Circuit Switched network, however, this is not the case. Each "connection" from codec #1 to codec #2 can, and usually does, take completely different paths through the network (based on issues that we are not concerned with for the purpose of this discussion). **This is where the first potential problem occurs.** Since the paths are physically different (and of different physical lengths), the first issue the terminals must deal with is related to the fact that data that is sent across each path may (will) take different amounts of time to actually propagate from one end to the other, depending on the physical path (and the possible processing points) between the two endpoints. Packets "leaving" one end at exactly the same time may (and do) arrive at different times at the far-end. Longer physical paths means data on those paths will take longer to get from one end to the other than data that transits on the shorter physical paths.

**In order to manage this problem**, and to provide a means by which data arriving on multiple paths at differing times can be realigned for proper decode and display, during the call setup sequence the terminals establish a Delay Table of sorts. This is a table that is calculated by defining the first path as reference "0", then measuring timing of data across subsequent paths to see if it arrives at "0" reference or "0 minus xxx ms [milliseconds] (late)" or "0 plus xxx ms (early)".

**Once the propagation delay for each path gets calculated by the terminals/codecs, these delay values are fixed for the duration of the call.** In the illustration above each terminal will, for instance, handle data on path or Channel #4 (the "orange" channel) as though it needs to be re-timed based on a "minus 5 ms" propagation delay. This means, of course, that as long as the proper delay is "accommodated-for" [factored into the data alignment calculations] on each Channel by each terminal, the arriving data can be properly re-aligned and properly decoded and displayed for the end-user. Elegant, fundamental and not-at-all processor intensive once calculated, this is a highly effective manner by which we handle the independent divergent paths for each Channel that get setup for a Circuit Switched/ISDN-based call and with-which we handle the data realignment.

## Anomalies in H.320 Videoconferencing, Section 4

**Now — let's step back for a moment:** In Circuit Switched telephony there is a function called "**Dynamic Rerouting**". This is a mechanism by which an individual "call" or connection through the network can be re-mapped or re-routed during the time that the call is actually active (during the time after call setup when end-user voice or video information is actually transiting the established connection). This re-routing was agreed to and is done in an effort to minimize the cost of the transport of the call to the telephone carrier company, and this was put in place with the understanding that the telephone service provider would be allowed to re-map or otherwise re-route calls dynamically in order to minimize their own costs if they agreed to #1 - not "drop" the call during this process and, #2 - share the resultant savings with their customers. In regular traditional analog voice communications (POTS or Plain Old Telephone Service) this Dynamic Rerouting is done all of the time, and is generally done many times during the period that a connection is active. Years ago you might have known this was happening by the pops and clicks that you would hear during your voice telephone call every time the connection was re-mapped. Today, with modern digital network systems and switches, this process is accomplished with virtually no noticeable changes or indications to the actual humans engaged in speaking across the connection. In and of itself, therefore, Dynamic Rerouting has helped to provide more affordable telecom services for you and I, the telephone service subscribers. It has been, and still is, "a good thing".

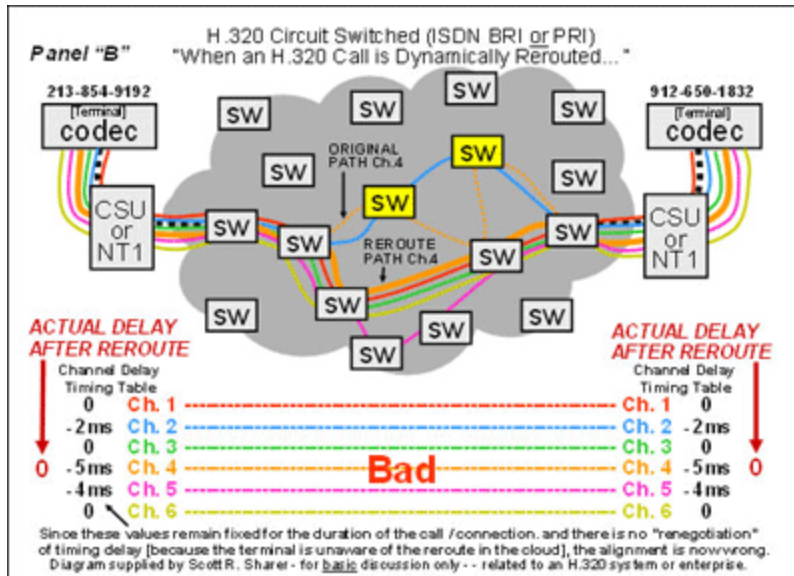
**Now** — Let's return to multi-channel Circuit-Switched Digital communications for H.320 videoconferencing.

**In Circuit Switched Digital connections for Videoconference data, we do NOT allow Dynamic Rerouting of the connection(s).** When we set-up or "Provision" the services for using Circuit Switching (H.320) communications in the digital network, we set a series of commands that actually tell the Local CO Services Provider switch, "Hey - I am an H.320 compliant videoconference digital encoder/decoder. I need 64kbps per channel; I need this-and-that-and-this-and-that, AND - for the purpose of handling my videoconference traffic across the network, make sure that every switch that gets engaged during the setup of this call knows that Dynamic Rerouting is to be turned "**Off**" for the channels that are used for transport of my voice and video data".

As already noted, this "statement" is actually set at the time that the account is set-up and the local switch is Provisioned (programmed) for our use in the Circuit Switched / ISDN network. Each time we go "off-hook" and enter a number string to place a call, our **SPID (Service Profile ID)** [in BRI] or our DID assignment [in PRI] alerts the **CO (Central Office)** switch "who we are" and "how it is to handle our connections" (how it is to set the port or ports in it's own switch-port frame, and what information it needs to pass along to every other switch that gets "engaged" and that will then handle our traffic for the duration of the call).

If we look at a modified form of the first Panel (Panel "A") that we originally looked at, and the discussion points that followed in this document, it becomes clear why we do not allow this very common Dynamic Rerouting function to occur with our H.320 videoconference calls / connections.

(Image Panel "B" shows the modification we mention and the resulting difficulty, allowing us to discuss a portion of the problem-set premise of this document at some length and begin to offer certain solutions).



**Let's discuss this very simple example of improper Dynamic Reroute.** Suppose that the "network" provider determines that the path or channel that was set-up for Ch. 4 (the orange channel) was necessary at the time the call was first set-up due to congestion in the other switches (used by Ch. 1, 3, and 6). Ch. #4 became, as a result of this chosen "path", the most "costly" means to carry this channel of information. Further suppose that, in the interest of achieving "least-cost" on the route, the network facilities had: #1- never been told that Dynamic Reroute was to be turned "Off", and #2 - had determined that that the initial congestion had been relieved and that a "lesser-cost" path was now available, similar to the path used for some other channels (1, 3 and 6). The facilities would then "Dynamically Reroute" that initial Ch. 4 "path" over to another shorter "path" in the [proper] interest of saving money.

**Problem? REMEMBER - The Delay Tables that were originally calculated by the terminals at the very start of the call were then "fixed".** Even though the new path results in "0 ms" delay, the terminals will continue to retime the data coming down Ch. 4 (the orange channel) with a "minus 5 ms" calculation. Now, instead of properly realigning the data, the terminals are actually actively mis-aligning the data, because they are working with an obsolete or incorrect Delay value. **Why don't the terminals simply "reset" the Delay structure?** Because - The Terminals are not aware that something like this has happened way out there in the network cloud. There is no mechanism for the switches in the cloud to report this to the terminals, nor is there a mechanism for the terminals to make queries of the various switches in the network related to issues such as this. Likewise - These are H.320 Standards-based terminals, and they "assume" that you and I have done the right thing and set the Provisioning for the network transport with Dynamic Rerouting turned "Off".

**What is the result of this now mis-aligned data?** There are two (2) most-common problems that will arise. First - The terminals will continue to decode the data, with portions of the data improperly aligned. These portions will often be "discarded" during the decode process, resulting in a loss of picture quality (tiling, jerkiness, "freezing" or loss of the image) or garbled audio. (Sounds like Problem #2 stated at the top of this discussion). **Second** - Though rare, it is entirely possible that the delay alterations will be so extreme, or that this will happen to more than one channel within a short period of time, so that the terminals may determine that there are unrecoverable synchronization errors on one or more channels in the connection, and they will act to defend the overall integrity of the call by ... you guessed it ... Downspeeding. (This sounds a lot like Problem #1 stated at the top of this discussion)

**So - What have we learned so far?** Most importantly we now know that we must make certain that the Provisioning for any terminal that is going to place / **initiate** a connection must include the setting that Dynamic Rerouting is to be turned "**Off**" for all channels that are engaged for our use in a Circuit Switched Digital ISDN H.320 videoconference call. The "answering" terminal does not have to do anything or have this Service Provisioning, since the structure and behavior of the network will be set by the terminal that is initiating, not the terminal that is answering the call. Obviously, however - Every system should have its' Network Switch Provisioning set with **Dynamic Reroute turned "Off"**, since systems are generally used to place as well as answer calls, and do this in a random manner. **Note:** This Provisioning also applies to Digital Network Services supplied to any Gateway or MCU that is ever going to place / **initiate** a call across an ISDN [or any other Circuit-Switched] connection to a terminal or to another Gateway or with / through an MCU.

**Circuit-Switched is Circuit-Switched** - In other words - the rules always apply for any Circuit-Switched H.320 connections (including ISDN BRI or PRI), whether the device that starts the call is an end-user VTC system or an intermediate piece of equipment or intermediate function.

## Anomalies in H.320 Videoconferencing, Section 5

Let's move on to two (2) additional complex issues:

Before we complicate the discussion image-panels and involve another possible "anomaly" in the network flow of H.320 Circuit-Switched digital videoconference traffic, we must first introduce two (2) additional concepts: **56k Restriction & Fallback to Telephony**. Though generally "known-about" by communication technicians, these are often not well understood, and people are frequently completely unaware of the potential these elements have to devastate a circuit-switched digital connection for use in H.320-based videoconference communications.

### Restricted Network

A "Restricted Network" [or network segment] is one that operates at 56kbps for all data, instead of 64kbps (64k Clear). Without going into a great deal of unnecessary math and background, **this is a leftover from the original Circuit Switched Digital Hierarchy in the North American Multiplex**. In the original NAM we made use of a system of Signaling with and through the network known as "RBS" or "Robbed Bit Signaling". This essentially means that in a 64k Channel (*we derive 64k by applying the Nyquist Criterion, Shannons Law and some other head-banging calculations to the conversions from the original analog telephone signal / channel in order to create a digital telephone channel*) we "signal" by robbing or stealing back some bits (actually resulting in 1kb for every 8kb or 8kb for every 64kb as a result of the 256 possible voltage levels) for the purpose of having bandwidth or "bits" to use to carry and convey signaling information in and for the network. Now, each of the 8-bits of the digital channel has multiple possible values, which are translated (by a codec) into the various possible voltages. Because the codec is designed for voice and video, these different voltages are not evenly spaced, making the closely spaced voltages more difficult to accurately decode. The voltage levels that a codec will use are referred to as the 'constellation'.

Using RBS, every time we reduce the constellation (# of voltage levels used) by 1/2, the available user bit rate for the link is reduced by 8k. (Hence - the "restriction".)

**In most computer communications and in two-way single channel voice telephone communications this results in no loss or problem to the end user.**

**Even in videoconference communications, having all of the channels operate for user-data at 56kbps instead of 64kbps will not result in any perceivable loss of video or voice quality in a multi-channel connection.** A 336k (Restricted 56) and a 384k (64k Clear) VTC connection will look and sound identical to the end-user.

**Problem(s)?** There are, of course, two (2) possible problems.

**First: It is possible that our terminal will be set to "speak" or send out data, and expect to "hear" or receive data, at 64k per channel** (most terminals, including Tandberg, Polycom and others, are set as a default to "communicate" at 64kbps per channel in a Circuit-Switched connection), **but will be connected to a network Service that operates only at 56k Restricted**. (Anyone dialing the old 700 numbers for AT&T Switched Digital H0 Service prior to AT&T adopting ISDN remembers operating each channel at 56k instead of 64k per second per channel).

**Second: It is possible that the local Service will be delivering 64k per channel to our terminal, and the remote terminal may also have 64k per channel arriving from their local Service Provider, but out there in the "Network Cloud" we may encounter or transit a portion of the network** (completely unbeknownst to us) **that operates only at 56k Restricted** (for a variety of reasons that do not matter to us in this discussion). Obviously, in any communications "system", every "point" from one end to the other must "speak" or "clock" at the same "rate". If every point does not, then there cannot be a single common "clock" and with no accurate common clock there can be no proper alignment of the data at each end prior-to and during decode.

**Now - "56k Restricted Network" is a "somewhat common" problem within North America, and is OFTEN a problem when dialing internationally from North America out to other countries** (and even when dialing an international call within North America from the United States out to Canada or to Mexico).

**Since, as we have stated above, the problem often occurs out in the "cloud", in places where we have no knowledge of the setup or configurations** (we don't even know what portion of the cloud our channels will move across - that is determined during call setup for each call / connection), **how do we know [determine] if we have**

terminals that are "talking and listening to 64k" but we have data that is moving across or are transiting through 56k Restricted segments or paths?

The symptoms of "56k Restriction" are very easy to recognize, and this problem should never cause anyone to lose valuable time sitting in a queue for Technical Support - it is **easily resolved [and immediately resolved]** by someone at the terminal or endpoint who is actually attempting to merely "dial" the call in the first place.

Please read and [if necessary] circulate to others or post the following so that people using your H.320 Circuit-Switched / ISDN - based VTC systems can better (more accurately and more quickly) determine that they have encountered a 56k Network Restriction problem with a particular H.320 Circuit-Switched videoconference connection, and so they are able to quickly "recover" from this difficulty and more forward with their meeting or conference.

**The Symptoms of 56k-Restrictions in a VTC H.320 Circuit Switched call: (some of the same symptoms may appear during an H.323 Packet Switched IP video call, but for very different reasons.)**

1. A call that connects on the first "B" or Bearer channel, negotiates, and as the remaining channels begin to be dialed / added / synchronized, the call immediately disconnects for no apparent reason. Audio and video data never "engage".
2. A call that connects, negotiates and aggregates all the channels properly, but shows only a black or blank screen for video (even though the terminal indicates it is sending and receiving video packets on all channels). Audio will usually be fine.
3. A call that connects, negotiates and aggregates all the channels properly, but shows only a frozen image on-screen for video (even though the terminal indicates it is sending and receiving video packets on all channels). Audio will usually be fine.
4. A call that connects, negotiates and aggregates all the channels properly, but shows only a black or blank screen for video (even though the terminal indicates it is sending and receiving video packets), then slowly begins to add video onto the screen, doing it tile-by-tile or block-by-block, like a huge random picture puzzle coming together. Once the complete received image is "filled-in" it is often frozen. Audio will usually be fine.
5. A call that connects, negotiates and aggregates all the channels properly, and shows any of the above video symptoms with screeching or howling audio (in telephony terms this is actually referred to as audio "Gritch" because it is manifest by a series of sharp sounds that sound a lot like someone saying the word "gritch" in a random pattern in a forced high-pitched voice). Some people call it audio "chirp".

**Now that we know the symptoms, what do we do about them? SIMPLE (and quick)**

1. **Hang up and dial the call again.** It is possible that the next time the call sets-up through the Circuit Switched network it will traverse different paths and any 56k Restricted path that was used before will not be used this second time.
2. In the "Call Quality" menu of your system, **set the terminal for "56 Restricted" before dialing** and then, when it places the call, it will negotiate connections through the network and with the far-end terminal or device at 56k per channel instead of 64k-Clear per channel.

**Note: Only the terminal placing or "initiating" the call needs to be set for 56k.** The "answering" terminal that is being "called" will automatically use the timing stated by the terminal that contacts it / places the call initially.

**ALSO Note: This "rule-set" of symptoms and solutions applies to Gateways and MCU devices as well as terminals / codecs. Any device that is placing calls across Circuit-Switched / ISDN connections can encounter this problem** and the "fix" or solution is always the same. **Always.**

**In certain locations and installations, this "56k-64k" mis-match can account for almost 1/3 [that's 33%] of all "problems" related to properly connecting ISDN Circuit-Switched H.320 VTC calls.**

**"Fallback to Telephony"**

This is a function that was introduced quite early on in the deployment of H.320 standards-based videoconference systems (in the late 1980's and into the early 1990's).

As we have previously noted, the early digital networks that provided end-to-end digital connectivity between or to end-user locations were quite brittle and prone to instabilities. We mentioned earlier that one attempt to provide a perception of higher stability in this environment was Downspeeding. In reality, Fallback to Telephony is an older mechanism. The thinking at the time (and this simple thought is true today) was that audio was and is far more important than video in distant communications. **You can have a meeting with audio and no video, but you cannot have it the other way around.** Given that people were used to audio conference-based meetings, the manufacturers configured a mechanism inside the videoconference codec that was and is known as "Fallback to Telephony". This command is designed to make certain that if anything goes "wrong" during the setup of a videoconference H.320 call / connection, then the system and the network should abandon the attempt at a multi-channel video-based connection and settle for a single channel audio-only voice telephone connection between locations. This meant that instead of "absolutely no way to engage in a meeting" if the video connection would not properly and fully complete, the end-user could count on at least having an audio conference connection, and though not nearly as effective as audio-plus-video, the participants could still talk and cover the important verbal discussion points of their meeting.

**NOTE: "Fallback to Telephony" actually has nothing to do with using a videoconference system to place or receive telephone calls to audio-only voice communication devices (telephones).** Even with no "Fallback to Telephony", or even with this turned "Off" in a videoconference endpoint, the system can still be used to place or receive a voice-only call. This is "FALLBACK"-to-Telephony, not "Enable / Allow" Telephony.

Though less important within the context of the more robust and stable Circuit Switched networks of today, this function is still part of every codec. In certain areas of the globe or in certain highly critical environments, this function remains quite valuable, enabling the end user to count on having "at-least" a voice communication.

**Problem(s) created by "Fallback to Telephony"?** Once again - there are two (2) basic problems with this function. **The first problem**, by itself, does not really represent a "problem", so let's call it an "issue" for now. The first "issue" is that this function is, by default, turned "On" in virtually every brand, model, make and type of system that has been or is being produced.

**The second problem**, when it rears its head, actually makes our previous "issue" into a problem. This second problem is created by virtue of how and what this function communicates to the network facilities in order to establish that, as stated above, if the multiple-channel video connection has "difficulties" during setup, it should abandon the attempt to engage multiple 64k channels in a 64-Clear trunk and map the call instead to a single channel on a voice-only trunk (in other words - it should "Fallback-to-'Telephony'").

**At this point**, let's take a look at a third image panel (Panel "C") and discuss how Fallback to Telephony can create multi-layered havoc in H.320 videoconferencing.



hang up and dial again (etc., etc.). **Or: In the worst possible case**, the terminal begins to see too much data that is now out of acceptable window for latency and will ... let's say it all together ... Downspeed the call.

**These two possible scenarios** sound a lot like Problem #1 and #2 as stated at the beginning of this document.

**What's really happening to cause all of this?** Simple. In the middle of the call, one of the switches in the path (indicated in yellow on image Panel "C") begins to get congested and it remembers that it was told that it was to handle your traffic at 64k-Clear with Dynamic Reroute turned "Off", but your codec also told it "...unless "something" goes "wrong"..." and made it difficult to maintain that data rate ... in which case it was told it could re-map your call, and even use a 56k trunk. It takes you at your word (your codec-passed command of "Fallback to Telephony" is invoked), it moves that connection (Ch. #4 - the orange line in Panel "C" above) to a 56k voice trunk, the timing falls apart in the Delay Table (just like before) AND - to make matters worst of all - you now also have 56k Restriction timing problems to deal with as well. **The one piece of good news?** Even if you and I are not happy about this result, the unknown and unidentifiable switch out there in the cloud is happy, because it was able to free-up congestion at that particular switch-point, and also conserve bandwidth in the network.

**So - What have we now learned about the impact of this feature command of "Fallback to Telephony"?** Most importantly we now know that we *must* make certain that the internal setup for any terminal that is going to place / *initiate* a connection must include the setting "Fallback to Telephony" set to "Off" for Circuit Switched Digital / ISDN H.320 videoconference calls. *ONCE AGAIN*— The "answering" terminal does not have to do a thing, since the structure and behavior of the network will be set by the terminal that is initiating, not the terminal that is answering the call. Obviously, however - Every system should have this set-to / turned "Off", since systems are generally used to place as well as answer calls, and they do this in a random manner, on a call by call basis.

*NOTE:* This "Fallback to Telephony" issue also applies to any and all Gateways or MCU's [they, too, have this setting inside in their setup menus] that are ever going to place / *initiate* a call across an ISDN connection (BRI or PRI) [or any other Circuit Switched connection] to a terminal or to another Gateway or MCU.

**Circuit-Switched is Circuit-Switched** - In other words - the rules always apply for any Circuit-Switched H.320 connections (including ISDN BRI or PRI), whether the device that starts the call is an end-user VTC system [a "codec"] or an intermediate piece of equipment or intermediate function [such as a GW or MCU].

## Anomalies in H.320 Videoconferencing, Section 6

### Final Summary

**While this document is not meant to convey that the possible difficulties we listed as "problem perceptions" at the start of the discussion can be traced back to these and only these specific root causes**, it is our considered opinion, within the context of what we have encountered as an ever-increasing occurrence over the last 10-14 months, that there is a very strong possibility that these discussion points will aid the reader in tracking down and further stabilizing the ISDN H.320 Circuit-Switched communications at or within their videoconference enterprise.

**In the interest of fuller discussion and in an acknowledgement of where most people begin to [improperly] focus when troubleshooting these problem symptoms**, we need to mention a couple of things related to any premature or uninformed decision that the problems have their root origin within either the Codec (VTC Terminal) or the Network (the Service supplied by the telephone company or supplied from an internal PBX device).

**There are many people who seize upon these other elements as causes for these types of symptoms, but these possible causes are extremely rare. Rest assured: The terminals or endpoints**, if they are up and running and stable for any period of time in any type of call, are not "failing" or "bad". There is no such thing as a codec that is "broken" for calls only to a certain city or country, or for long-distance not local calls. If it can reliably make local or LD calls to one city, it can do it to any city - there is something else that is causing the difficulty.

**The Network Services provider**, if able to provide certain stable connections at all are generally not to blame for the instability - there is no such thing as a Network Service that works 100% for talking to this endpoint or that location but not all of the time. Something is getting lost or "confused along the way" under certain conditions in the network.

**WARNING -- Whenever you encounter the problems described at the beginning of this document**, if you start out by narrowly focusing on the terminal or the network and steadfastly maintain that either the codec manufacturer or the network service provider [or both] are not being entirely forthcoming, then you will often experience months and months of having to chase-your-own-tail while never solving the problems.

**Additionally ...**

**DO NOT OVERLOOK THE IMPACT OF THE STATE OF NETWORKS IN OTHER PARTS OF THE GLOBE ON THE QUALITY AND STABILITY OF YOUR COMMUNICATIONS.**

When it comes to the "maturity" of physical network and services, some places on earth are far more advanced than we are in the United States. *Most are far less advanced.* This author traveled to one particular country back in 1999 (a country that is not even considered to be a "Developing" or 3rd-World country) and found ISDN being run *over barbed-wire.*

This had an "undesirable and disappointing impact" when connections from the United States were passed along that set of "wires". And, by the way - the power in those areas is as dirty as the network [if not more-so], and, since we derive "clock" from the power (50Hz or 60Hz), unstable / "dirty" power creates... you guessed it... unstable clock.

*A couple of additional "examples" of "creatively deployed physical plant":*



(If I hadn't seen it myself - - Difficult to imagine or believe...but true)

**Even in modern digital environments like here in the United States - -ALSO NOTE - Any and every Gateway or MCU Service Provider or Gateway or MCU functional device that is going to be a part of any portion of any connection will have to be assessed,** as will any device that is placing calls into meetings and connecting with other properly configured Service and endpoint.

**And Remember - Errors in timing in an MCU can and generally do propagate across all of the ports in the conference.**

All of those "stable" terminals and connections in your enterprise or within the enterprise of companies you are going to communicate with that show no problems and have reliable and proper configurations may suddenly develop the symptoms we have discussed as a result of the anomalies introduced by just one (1) of the terminals in a multi-point connection.

### **In Closing**

I appreciate the opportunity to share this information with each of you reading this document, and I sincerely hope that it acts to shed some light on the issues that you may be confronting. It is our hope that stepping back, taking stock of some of the fundamentals and making the small adjustments we have detailed here will go far towards providing strong, stable and high quality video communications for your ISDN-based / Circuit-Switched H.320 systems well into the future.

Respectfully Submitted:

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## Anomalies in H.320 Videoconferencing, Section 7

### Background: ISDN and Circuit Switched Network Configuration

**ISDN, or Integrated Services Digital Network, is a suite of internationally-adopted standards for end-to-end digital communication over the public telephone network.** As this network has been traditionally oriented towards voice services, ISDN was an important step forward in the adaptation of the network to handle the increasing global demand for computer-to-computer data communications. ISDN brought us closer to the goal of a ubiquitous multi-service network, integrating voice, data, video and image services in a digital format over a common, global network.

While ISDN differs fundamentally from conventional telephone service (called Plain Old Telephone Service or POTS), ISDN has been designed to allow end-to-end compatibility for voice services. Voice calls can be made to or from an ISDN line from a POTS line anywhere in the world. Equally important is the fact that ISDN service can be carried over the existing telephone network infrastructure. This infrastructure represents a massive global investment over the past century in central office switches, which route calls and handle billing; in transmission systems which carry the large volume of calls to remote destinations, largely over optical fiber cable today; and millions of miles of twisted-pair copper cabling to carry services to our homes and offices.

While ISDN standards work is continuing on even higher data-rate services such as full-motion video, two well-defined ISDN interface standards are in common use today:

**Basic Rate Interface (BRI)** - The term ISDN or ISDN line is often used synonymously with the Basic Rate Interface. The Basic Rate Interface defines a digital communications line consisting of three independent channels: **two Bearer (or B) channels, each at 64 Kilobits per second, and one Data (or D) channel at 16 Kilobits per second.** For this reason the ISDN Basic Rate Interface is often referred to as 2B+D.

**The B channels are used for carrying the digital information,** whether computer data, digitized voice, or motion video with appropriate equipment these B channels can be bonded or linked together to provide an aggregate 128 Kilobits per second data channel. The D channel is used to carry signaling and supervisory information to the network, and can also be used to carry packet-mode data over an X.25 network.

Each of the two B channels is treated independently by the network, allowing for simultaneous voice and data, or data only, connections to different locations. With specialized hardware and software, multiple B channel connections can be aggregated to achieve file transfer rates of several Megabytes of data per minute or more.

**Primary Rate Interface (PRI)** - The Primary Rate standard is a higher-level network interface defined at the rate of 1.544 Megabits per second (for North America). This particular rate was selected for compatibility with T1 digital lines commonly used today. The Primary Rate is comprised of **23 B channels, each at 64 Kbps, and one 64 Kbps D channel for signaling.** These B channels can interconnect with the Basic Rate Interface, or when carrying voice services to any POTS line.

### Wiring Standards - BRI or Basic Rate Interface

The reason for discussing standards is to increase awareness and promote adherence in practice. This section contains a summary of the wiring standards applicable to ISDN. Refer to EIA/TIA-570 Residential and Light Commercial Telecommunication Wiring Standard for more details. This standard specifies the number of conductors and cable quality. Cables for new installations should contain four twisted pairs of solid conductors. The additional pairs will support other functions, such as DC power distribution to TEs. The standard recommends 8 position modular jacks with 8 conductors. Choose either T568A or T568B jacks and use throughout the building. The TSB-40 standard defines five levels or "categories" of cable characteristics. Category 3 cable meets the requirements for new installation of telephone and ISDN services. Installing Category 4 or Category 5 cable may save money in the long run if future higher rate services are expected. The standards specify Star wiring to each service outlet from the distribution device. Although daisy chaining outlets in the same room is allowed by the standard, this is not recommended. The length of cable from the distribution device to each service outlet shall not exceed 90 meters (295 feet).

**CAUTION: We do not recommend that wiring leave the building.** If it does, then consult your local codes for additional protection requirements. Cables exposed to the elements may degrade.

## **Distribution Device**

The distribution device required by EIA/TIA 570 allows multiple outputs for a single input. Such a device can support POTS as well as ISDN. Wires placed over insulation displacement connectors in multiples of 4 make connections by snapping a cap onto the connector. No special tools are needed. This device can support star wiring (multiple outputs for a single input) to multiple target rooms by connecting the ISDN signal to several input rows. For ISDN in 1 target room, only 1 output is needed. Use a 24 gauge Category 3 cable or better to connect the ISDN U signal to the target room.

## **Modular Jacks**

When installing 4 pair cables, the modular jacks must terminate all pairs. The acceptable modular jacks conform to T568A or T568B wiring. These 8 position modular jacks contain 8 conductors. EIA/TIA categories also apply to modular jacks. The minimum requirement is Category 3 jacks when rewiring or in new installations. The standard recommends matching the cable and jack categories, however a Category 3 jack used with a higher Category cable provides improved performance for ISDN over a Category 3 cable and allows for easier system upgrade.

Modular jacks for ISDN differ from the typical POTS jack. Color code or label the jack with "ISDN U" to eliminate confusion and prevent connecting incompatible equipment to the jack. The Category 3 connecting cords used are also different than ordinary phone cords.

## **NT1 in the Target Room**

The NT1 and a power source will also reside in the target room. Some supplies can power the ISDN TE as well. Only 1 pair is used to carry the ISDN U signal to the target room. Centrally locate the distribution device for the possible future conversion to multiple target rooms. Note that some terminal equipment may have an integrated NT1.

## **Wiring Summary**

A summary of minimum equipment requirements for rewiring according to the standards:

- **Category 3 or higher cables and suitable connecting cords.**
- **Connecting cords should not exceed 10 feet.**
- **Category 3 or higher modular jacks** intended for T568A or T568B wiring.
- A distribution device (not essential for this application, but required for full standards compliance and recommended for flexibility).

The next sections describe the tasks necessary to install/verify the ISDN service from the Demarc device to the target room.

### **Task - Wiring from the Demarc to the Target Room**

Plan the route from the Demarc to the target room to determine the length of cable needed. To avoid shock, do not connect the cable to the legal Demarc yet. If you are replacing an existing service, you may not have a legal Demarc point. Take care when fastening cable to the building not to crush or puncture the cable.

### **Using a Distribution Device**

Mount a distribution device in a suitable location, preferably centralized to minimize the wire runs for future wiring. Even though the ISDN service uses 1 pair, connect all 4 pairs to the distribution device if it accepts 4 pairs, according to the color code. Run the cable to the target room, leaving some slack at the distribution

### **Task - Installing the 8 Pin Modular Jack**

Once the wiring reaches the desired location, strip back the outer jacket and connect all 4 pairs to the 8 pin modular jack according to the color code. Mount the jack to the wall or baseboard and place the cover plate on the jack. You should now have a continuous pair from the Demarc to the target outlet. Label this outlet "ISDN U" to avoid connecting the wrong equipment. Now connect pair 1 of the cable coming from the distribution device (if present) to the ISDN service at the legal Demarc.

### Connecting Equipment in the Target Room

Now connect your equipment to the ISDN jack. Consult the guidelines from your equipment manufacturer.

**The NT1 should provide a 50 Ohm resistance termination.** This 50 Ohm termination can be provided in many ways depending on the NT1 design. Some vendors provide a 50 Ohm terminating resistor, others may provide a combination of terminators that achieves a 50 Ohm termination. See the manufacturer's instructions for proper NT1 optioning.

If your NT1/NT384 has a timing option, **choose the default.**

Connect the power supply to the ISDN modular jack and the NT1 to the power supply. Then connect the ISDN terminal to the NT1. **Use Category 3 or better cords.** Plug the power supply into an AC outlet. Program your ISDN TE with the appropriate parameters (e.g. DNs, SPID, etc.) following the manufacturer's instructions. Certain parameters (e.g., DNs, SPID) are provided by your service provider. If your service includes voice, you should now be able to make a call from your ISDN TE. If you ordered data service, then the terminal instructions should guide you through the options needed to establish data calls. Once the power supply is plugged into AC, you may notice indicators lighting on the NT1 that show line and equipment status.

### PRI: What to order or configure

Specify the following requirements when you order connectivity and services:

- 23 B-channels + 1 D-channel
- D-channel on channel 24 (timeslot 24). Do **not** order NFAS (non-facility associated signaling service), which enables you to use channel 24 as a data-carrying B-channel.
- **Layer 1 line code is B8ZS** [not AMI] **with ESF** (Extended SuperFraming)
- Data rate of 1.544 Mbps
- **64 kbps clear channel** service which ensures calls will not be routed over 56 kbps channels
- **Companding** type is **μ-law**
- A **CSU** is required
- The **T1 interface number** must be **0**
- The **D-channel** must be specified as the **terminal endpoint identifier (TEI) 0**.
- **If the switch type is AT&T/Lucent**, request that allocation of channels for incoming calls is in descending order, high to low (23 to 1).
- **REMEMBER** – If using a PBX in the connection flow, the **Hunt direction must match between the CO settings and the local PBX**
  
- **ALSO Note:** A Tandberg 6000 or 6000MXP codec **requires** that the **Delta** or "D" channel **make appearance on the 24th channel for PRI**, and...

... **all Dynamic Rerouting must be turned "Off"** in the terminals and in the switch (PBX or CO Switch). In the terminals, this includes turning the function of "Fallback to Telephony" to "Off" (default in a Tandberg and in most other major brands is "On")

### BRI: What to order or configure

ISDN is a switched network capability that supports services that are designed to provide end-to-end digital connectivity for the simultaneous transmission of voice and data. Basic Rate ISDN service consists of a 2-wire line port associated with a local switching system, a 2-wire Central Office Distributing Frame (CODF) cross-connect

between a 2-wire CLEC CODF termination and the BRI CODF termination. At each 2-wire interface, one conductor is called tip and the other conductor is called ring. **The BRI interface provides Line Termination (LT) functionality and utilizes the Two-Binary One-Quaternary (2B1Q) line code** operating at 160 Kbps (192kbps with the H0 overhead).

**BRI is available in the 2B + D configuration** which provides two B channels and one D channel (for signaling).

**Request from the services provider a National ISDN 1 or 2 ISDN-BRI line with 2B1Q line code.** Your ISDN line must be configured to **allow circuit switched voice and data on both B-channels** and signaling on the D-channel. **(This is the essential configuration for BRI if you reference the IOC code Package "S" - see the complete list of packages beginning on page 25 in this document).** The telephone company should supply you with a separate telephone number and SPID (Service Profile Identification) for each B-channel; however, it will physically be only one ISDN line. Request that the Telephone Company program your ISDN line with the following attributes: B1 and B2 should be set as follows:

- Set the **circuit switch option to Yes**; set the **barrier restriction option to no packet mode data (NO\_PMD)** only (this tells the switch that you require circuit switch ability on your B-channel, The bearer restriction on your line means that you are not allowed packet data on your B-channel.)
- Set **protocol to function version 2**; (PVC2) (this tells the switch that your CPE software is using National ISDN 1 or 2 protocol.)
- Set the **service profile identification (SPID) suffix**. The SPID format is generally going to be (with modern services and switches) area code + 7 digit number + 1 + 00. **If the CO switch is a Nortel DMS100 or 200 series, this is definitely the SPID format structure.**

#### **Format and Definition of a National ISDN SPID**

For National ISDN-1 or 2 (NI-1 / 2) and beyond, a SPID is a free-formatted numeric string between 3 to 20 characters in length. The SPID is assigned by the Local Exchange Carrier or LEC (ISDN service provider) when the customer places a service order for a BRI. Currently, **the customer must enter the SPID into the terminal before the terminal can initialize at layer 3.**

The **Service Profile Identifier (SPID)** is a number assigned to a fully initializing ISDN terminal that enables the Stored Program Control Switching System (SPCS) to **uniquely identify the ISDN terminal at layer 3 of the D-channel signaling protocol.**

Each fully initializing terminal needs to be uniquely identified at layer 3 in order for the SPCS to provide terminal-dependent features. When the terminal sends its SPID during layer 3 initialization, the SPCS uses the SPID to associate the terminal with a specific set of terminal services (that have been previously provisioned in the SPCS for that terminal). The directory number of the terminal is not sufficient for this purpose because directory numbers may not uniquely identify a terminal on an interface; that is, a directory number can appear on multiple terminals on a single BRI.

**SPIDs also simplify switch-based troubleshooting** when multiple terminals are assigned to a BRI. **Abnormality logs in the SPCS indicate the SPID of a terminal experiencing a problem.** This information helps maintenance personnel isolate and resolve ISDN troubles.

#### **One example of a configuration where different terminal-dependent features need to be assigned follows:**

Two voice terminals are served by a single BRI. The terminals share the same directory number, but only one terminal is provided with access to call hold and flexible calling (conference calling) features. A separate set of characteristics--or profile--is provisioned in the SPCS for each terminal. One profile includes call hold and conference calling; the other profile does not. When each terminal initializes, the SPCS uses the terminal's SPID to provide the correct features and button assignment to the terminal. There are many other examples of features that are assigned in a similar manner.

- Set the **Terminal Endpoint Identifier (TEI) to Dynamic** (this tells the switch that you can accept any TEI value from 64 to 126.)

- Set **Ring to Yes** (this tells the switch to send an alerting message to your CPE when there is an incoming call.)
- Set **Key system (EKTS) option to No** (this tells the switch that you are not a key system.)
- Place the **lower layer compatibility option for data on the B-channels** (this tells the switch that your CPE may utilize the lower layer compatibility information element for data on the B-channels.)
- Set the **Lower Level Provisions to ensure that Dynamic Rerouting is set to "Off."** In the terminals, this includes turning the function of "Fallback to Telephony" to "Off" (default in a Tandberg and in other major brands is "On")

If you are actually using the lines terminated in such a manner as to have multiple devices active on a single BRI line service at the same time\*, then:

- Place calling subaddress option for data on the B-channels (this tells the switch that your CPE will send a subaddress.)
- Place called subaddress option for data on the B-channels (this tells the switch that your CPE can receive a subaddress.)

**PLEASE NOTE:** The use of multiple terminal devices on a single terminated ISDN BRI line, if one of the terminal devices is an H.320 videoconference terminal, is **NOT recommended** (nearly every terminal on the market does not like to "share" a BRI line with other active devices)

**National ISDN Ordering Codes (IOCs)** have been adopted by the Regional Bell Operating Companies to facilitate ISDN ordering and line provisioning. IOCs can be divided into two categories--NI-1 IOCs and NI-2 IOCs.

**Generic IOCs, or ISDN Capability Packages,** reflect a standard set of commonly used interface configurations for an ISDN Basic Rate Interface.

**Non-Generic IOCs** are based on Generic IOCs. The Industry recommends Generic IOCs be used wherever possible. However, where this is not possible, Equipment/Applications Developers may register a Non-Generic IOC with Telcordia.

**Archived IOCs:** The North American ISDN Users' Forum (NIUF) has approved a new methodology to further streamline the IOC simplification process - IOC archiving. Many Generic and Non-Generic IOCs have had very limited or no activity over the last few years. Also, many Generic IOCs have never had CPE registered with them. IOCs that fall into this category have been archived. Archived IOCs are noted in the following Generic IOC listing.

The definition of an archived IOC is as follows:

- The National IOC Process will not register any new equipment with an Archived IOC.
- Service providers will not take any new orders for an Archived IOC.
- Operational support will not be provided for an Archived IOC.
- An Archived IOC will remain in Telcordia documentation, but it will be flagged as Archived
- The letter designation for an Archived IOC will not be reused.
- An Archived IOC can be reactivated at a later date, if desired. However, operational support may be needed to bring it up to date.  
Note that a Non-Generic IOC continues to be valid, even if the Generic IOC it was based on has been Archived.

**EZ-ISDN IOCs** are designed as "superset" packages that can simultaneously meet the needs of many of the top volume mass market applications. CPE may register as either Fully Conformant or Compatible with an EZ-ISDN IOC.

A key aspect of the EZ-ISDN IOCs is that they offer a more "relaxed" registration criterion for registration of CPE. This relaxed registration enables a more varied set of CPE to register with the EZ-ISDN IOCs, making ISDN easier for the mass market to order. The "relaxed" registration criterion is known as Compatible. Whereas CPE must be Fully Conformant with (exactly match) the Generic and Non-Generic IOCs (e.g., Capability S), the Compatible criterion allows CPE to match only some of the features and capabilities in the EZ-ISDN IOCs. For CPE to register as Compatible with the EZ-ISDN IOCs, the CPE must work on a line provisioned with the EZ-ISDN IOC, although it may not necessarily make use of all the features and capabilities defined in the EZ-ISDN IOC. Also, the features and capabilities that the CPE does not utilize, but are still part of the EZ-ISDN Translations, must not adversely affect the operation of the CPE.

Note: All switch translations associated with an EZ-ISDN IOC are programmed into the ISDN switch. It is the CPE that has the option to use some or all of the translations.

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### **Generic ISDN Ordering Codes - a.k.a. National ISDN Capability Packages**

Capability Packages A through Q were developed by the North American ISDN Users Forum (NIUF) and implemented in conjunction with the industry's ISDN Solutions '94 event. Since that time, additional Capability Packages have been developed with Telcordia. The National ISDN Registry of Customer Equipment lists equipment models that have enrolled in the NIUF-supported registration process provided by Telcordia. Contact your service provider for ISDN Capability Package pricing and ordering information for your home or office.

#### **Capability Package Definitions**

**Capability Package A:** Basic D-Channel Packet services. No voice capabilities are provided.

**Capability Package B:** Circuit Switched Data on one B-Channel. No voice capabilities are provided, basic voice capabilities (no features) are supported.

**Capability Package C:** Alternate Voice/Circuit Switched Data on one B-channel. Only basic voice capabilities (no features) are supported.

**Capability Package D:** Voice on one B-Channel and basic D-Channel Packet services. Only basic voice capabilities (no features) are supported.

**Capability Package E:** (ARCHIVED) Voice on one B-Channel and basic D-Channel Packet services. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer.

**Capability Package F:** (ARCHIVED) Voice on one B-Channel and basic D-Channel Packet services. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer, and uses CACH EKTS.

**Capability Package G:** Voice on one B-Channel and Circuit Switched Data on the other B Channel. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer.

**Capability Package H:** (ARCHIVED) Voice on one B-Channel and Circuit Switched Data on the other B Channel. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer, and uses CACH EKTS.

**Capability Package I:** Circuit Switched Data on both B-Channels. No Voice or Packet capabilities are provided.

**Capability Package J:** Alternate Voice/Circuit Switched Data on one B-Channel, and Circuit Switched Data on the other B Channel. Only basic voice capabilities (no features) are provided.

**Capability Package K:** Alternate Voice/Circuit Switched Data on one B Channel, and Circuit Switched Data on the other B-Channel. Voice capabilities include Three-way (Conference) Calling Call Hold, Call Drop, and Call Transfer.

**Capability Package L:** (ARCHIVED) Alternate Voice/Circuit Switched Data on one B-Channel, and Circuit Switched Data on the other B Channel. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer, and uses CACH EKTS.

**Capability Package M:** Alternate Voice/Circuit Switched Data on both B Channels. Only basic voice capabilities (no features) are provided.

**Capability Package N:** (ARCHIVED) Alternate Voice/Circuit Switched Data on one B Channel, Circuit Switched Data on the other B Channel, and D-Channel Packet. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer.

**Capability Package O:** (ARCHIVED) Alternate Voice/Circuit Switched Data on one B Channel, Circuit Switched Data on the other B Channel, and D-Channel Packet. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer, and uses CACH EKTS.

**Capability Package P:** Alternate Voice/Circuit Switched Data on both B Channels, and D-Channel Packet. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer.

**Capability Package Q:** Alternate Voice/Circuit Switched Data on both B Channels, and D-Channel Packet. Voice capabilities include Three-way (Conference) Calling, Call Hold, Call Drop, and Call Transfer, and uses CACH EKTS.

**Capability Package R:** Circuit Switched Data on 2 B channels. Data capabilities include Calling Number Identification. No voice capabilities are provided.

**Capability Package S:** RECOMMENDED FOR VTC H.320 CODEC Alternate Voice/Circuit Switched Data at 64kbps per Channel on 2 B channels. Data and voice capabilities include Calling Number Identification.

**Capability Package T:** (ARCHIVED) Voice on 2 B channels and basic D-channel packet. Only basic voice capabilities are provided, with no features.

**Capability Package U:** Alternate Voice/Circuit Switched Data on both B channels. Voice capabilities include non-EKTS voice features including Flexible Calling, Call Forwarding Variable, Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery). Data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). This package is identical to EZ-ISDN1 with respect to switch translations.

**Capability Package V:** Alternate Voice/Circuit Switched Data on 2 B channels. Voice capabilities include non-EKTS voice features including Flexible Calling, Advanced Call Forwarding (i.e., Call Forwarding Variable, Call Forwarding Interface Busy, Call Forwarding Don't Answer, and Message Waiting Indicator), Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery). Data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). This package is identical to EZ-ISDN1A with respect to switch translations.

**Capability Package W:** Capability W is equivalent to Capability R with the addition of D-channel packet capabilities. Capability Package W includes circuit-switched data on two B-channels, and D-channel packet. Three Directory Numbers are provisioned; each circuit-switched Directory Number is allowed access to only one B-channel at a time (unlike Capability Package R). Circuit-switched data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). No voice capabilities are provided.

**Capability Package X:** Capability X is equivalent to Capability S with the addition of Additional Call Offering for voice, as well as D-channel packet capabilities. Capability Package X includes alternate voice/circuit-switched data on two B-channels, and D-channel packet. Three Directory Numbers are provisioned; each circuit-switched Directory Number is allowed access to only one B-channel at a time (unlike Capability Package S). Voice features include Additional Call Offering and Calling Number Identification (which includes Redirecting Number Delivery). Circuit-switched data capabilities include Calling Number Identification (which includes Redirecting Number Delivery).

**Capability Package AB:** Capability Package AB is equivalent to Capability Package U, with two changes: the inclusion of all the voice features on both Directory Numbers and the addition of D-channel packet capabilities.

Capability Package AB includes alternate voice/circuit-switched data on two B channels and packet data on the D-channel. Three Directory Numbers are provisioned; each circuit-switched Directory Number is allowed access to only 1 B channel at a time. This package provides non-EKTS voice features, including Flexible Calling, Call Forwarding Variable, Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery) on both circuit-switched Directory Numbers. Circuit-switched data capabilities include Calling Number Identification (which includes Redirecting Number Delivery) on both circuit-switched Directory Numbers.

**Capability Package AC:** Capability Package AC is equivalent to Capability Package V, with two changes: the inclusion of all the voice features on both Directory Numbers and the addition of D-channel packet capabilities. Capability Package AC includes alternate voice/circuit-switched data on two B channels and packet data on the D-channel. Three Directory Numbers are provisioned; each circuit-switched Directory Number is allowed access to only 1 B channel at a time. This package provides non-EKTS voice features, including Flexible Calling, Advanced Call Forwarding (i.e., Call Forwarding Variable, Call Forwarding Interface Busy, Call Forwarding Don't Answer, and Message Waiting Indicator), Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery) on both circuit-switched Directory Numbers. Circuit-switched data capabilities include Calling Number Identification (which includes Redirecting Number Delivery) on both circuit-switched Directory Numbers.

## **EZ-ISDN IOCs**

**EZ-ISDN 1** includes alternate voice/circuit-switched data on two B-channels. This package provides non-EKTS voice features, including Flexible Calling, Call Forwarding Variable, Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery). Data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). This package is identical to Capability Package U with respect to switch translations.

**EZ-ISDN 1A** includes alternate voice/circuit-switched data on two B-channels. This package provides non-EKTS voice features, including Flexible Calling, Advanced Call Forwarding (i.e., Call Forwarding Variable, Call Forwarding Interface Busy, Call Forwarding Don't Answer, and Message Waiting Indicator), Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery). Data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). This package is identical to Capability Package V with respect to switch translations.

**EZ-ISDN 2** includes alternate voice/circuit-switched data on two B-channels. This package provides CACH EKTS including two secondary directory numbers, and voice features, including Flexible Calling, Call Forwarding Variable, and Calling Number Identification (which includes Redirecting Number Delivery). Data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). Note that Additional Call Offering functionality is incorporated in the EKTS service. This package does not have a corresponding Capability Package with a letter designation.

**EZ-ISDN 2A** includes alternate voice/circuit-switched data on two B-channels. This package provides CACH-EKTS including two secondary directory numbers, and voice features, including Flexible Calling, Advanced Call Forwarding (i.e., Call Forwarding Variable, Call Forwarding Interface Busy, Call Forwarding Don't Answer, and Message Waiting Indicator), and Calling Number Identification (which includes Redirecting Number Delivery). Data capabilities include Calling Number Identification (which includes Redirecting Number Delivery). Note that Additional Call Offering functionality is incorporated in the EKTS service. This package does not have a corresponding Capability Package with a letter designation.

**EZ-ISDN 3** is equivalent to EZ-ISDN 1, with the addition of D-channel packet capabilities. EZ-ISDN 3 includes alternate voice/circuit-switched data on two B channels and packet data on the D-channel. Three Directory Numbers are provisioned; each circuit-switched Directory Number is allowed access to only 1 B channel at a time. This package provides non-EKTS voice features, including Flexible Calling, Call Forwarding Variable, Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery). All these voice features, with the exception of Calling Number Identification, are available on only the first circuit-switched Directory Number; Calling Number Identification is available on both circuit-switched Directory Numbers. Circuit-switched data capabilities include Calling Number Identification (which includes Redirecting Number Delivery) on both circuit-switched Directory Numbers.

**EZ-ISDN 3A** is equivalent to EZ-ISDN 1A, with the addition of D-channel packet capabilities. EZ-ISDN 3A includes alternate voice/circuit-switched data on two B channels and packet data on the D-channel. Three Directory Numbers

are provisioned; each circuit-switched Directory Number is allowed access to only 1 B channel at a time. This package provides non-EKTS voice features, including Flexible Calling, Advanced Call Forwarding (i.e., Call Forwarding Variable, Call Forwarding Interface Busy, Call Forwarding Don't Answer, and Message Waiting Indicator), Additional Call Offering, and Calling Number Identification (which includes Redirecting Number Delivery). All these voice features, with the exception of Calling Number Identification, are available on only the first circuit-switched Directory Number; Calling Number Identification is available on both circuit-switched Directory Numbers. Circuit-switched data capabilities include Calling Number Identification (which includes Redirecting Number Delivery) on both circuit-switched Directory Numbers.

### **Non-Generic IOCs**

Equipment/Applications Developers interested in registering Non-Generic IOCs should obtain Telcordia publication SR-3480, "Procedures for Performing an IOC Translations Review." This document contains a set of guidelines that companies must follow to ensure that a consistent set of documentation is acquired for all registered IOCs. Telcordia will provide a contract to the registering company along with a statement of Telcordia services and registration fees. The set of switch translations needed to correctly configure an interface to a customer's ISDN equipment/application are obtained by conducting Translations Reviews in a lab setting, also available through Telcordia. All Telcordia documents may be ordered on-line or call 1-800-521-2673.

### **Additional Telcordia publications describing the IOC process and content:**

SR-3622, *National ISDN Ordering Codes Overview and Code Listing*, lists all Generic/Non-Generic IOCs and the CPEs that are registered with them to date. SR-3622 is intended for industry stakeholders that need information on IOCs and Customer Premise Equipment (CPE) but not the actual switch translations. SR-3622 may be ordered on-line or call 1-800-521-2673.

*National ISDN Ordering Codes*, plus revisions, informs the industry of the National IOCs that have been registered to date with Telcordia, including the switch translations for all Generic and Non-Generic IOCs plus their associated CPE and IOC Fact Sheets. *National ISDN Ordering Codes* is intended for industry members, most particularly service providers (LECs) with responsibility for service negotiation, service orders and recent change/memory administration.

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