Kofax Communication Server

TCOSS ISDN Technical Manual

Version: 10.2.0

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1. ISDN Information

ISDN, abbreviated form of **I**ntegrated **S**ervices **D**igital **N**etwork, is a set of ITU-T (formerly known as CCITT) standards for digital transmission over ordinary telephone copper wire as well as over other media. These protocols are accepted as standards by virtually every telecommunications carrier all over the world.

Integrated	Because Telecommunication and Data communication on one system
Services	Voice, data, text, picture, video, telex, facsimile, telemetry alarms and more
Digital	End to End Digital Transmission with voice digitization handled in the terminal
Network	Worldwide network based on existing telephone lines, providing standard interfaces and call procedures.

The essential difference between ISDN and the conventional telephone system is that it is digital not analogue. Information travels as bits rather than as waves. In addition, it also allows multiple streams of these bits to occupy the same connection, providing the user with greater versatility of services.

The Integrated Services Digital Network uses the twisted-pair copper telephone line that would traditionally carry only one voice connection. ISDN can carry more than one connection over this wire at the same time, and at greater speed. Applications include telecommuting; simultaneous voice, fax, data and email; inexpensive videoconferencing; remote broadcasting and high quality audio transmission.

Important! The Kofax Communication Server and its components formerly used the name TOPCALL. Some screen shots and texts in this manual may still use the former name.

1.1 Digitizing Voice and Pulse Code Modulation

To carry human voice in a digital form, the voice signal is sampled 8000 times per second. This is a result of the so called “sampling theorem” which states that to be able to accurately reproduce an analogue signal from a series of samples, sampling must occur at twice the highest frequency of the signal.

The human voice passband is taken to be between 0 and 4000 Hz to provide an appropriate guard band for protection between adjacent voice channels when multiplexing is performed. The maximum frequency, 4000 Hz, requires a sample rate of 8000 times per second!

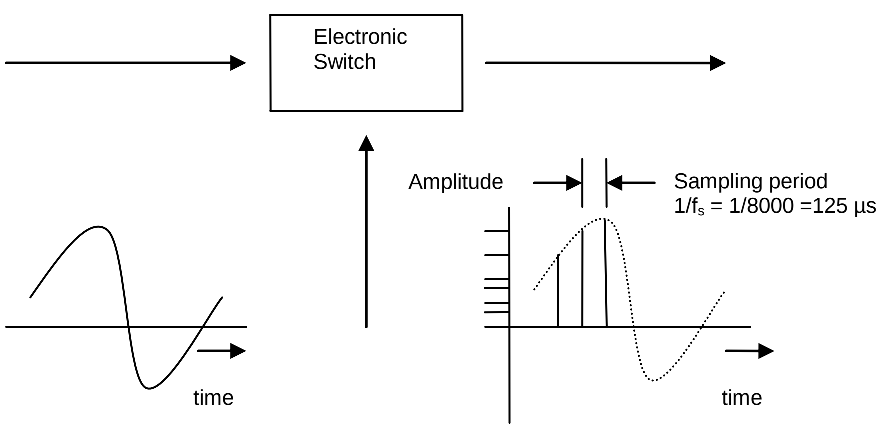
Note: $f_s > 2 \times f_m$

f_s … Sampling frequency
f_m … Maximum frequency

Each sample of the voice signal is converted into a digital bit stream. The process of converting the analogue sample into a bit stream is Pulse Code Modulation, short PCM and is performed by a device called a CODEC (Coder DECoder).

The drawing below shows the voice digitalization scheme.

Sampling of voice



The voice signal is sampled every 125 µseconds or once every 1/8000 second. This sampling is called Pulse Amplitude Modulation (PAM). Please note the non-linearity of the PCM amplitude scale above. The defined amplitude levels are closer together at lower volumes and farther apart at higher volumes. This is called “companding”.

A non-linear scale is used because it is more important to catch the subtleties and nuances when people are talking softly than when they are yelling. There are two main companding schemes used in digital telephony (A-Law and µ-Law):

Due to the fact that certain signal types (e.g.. analogue voice) have an infinite number of “States” , where it would appear on an amplitudinal scale, Quantizing was conceived. The digital quantizing scale has only 255 values. 0 to +127 and 0 to -127, an 8 bit value is assigned to the state of the non-digital signal. The scale differs in the µ-Law and A-Law companding algorithms.

µ-Law which is used in North-America and Japan (15 Segments), where graduations are closer at the lower amplitudes and increase in difference as the amplitude rises.

A-Law which is used in all other countries, also Europe (13 Segments), where the graduations are more uniform.

1.2 Quantization and Encoding

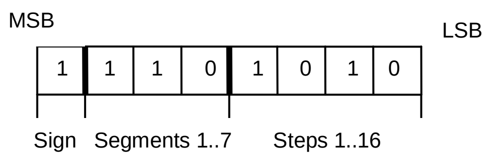
In the E1 system, the PAM signals are quantized using a 13-segment compression (15-segment within T1 known as µ-Law) characteristic known as the A-Law.

This characteristic is made up of 7 different size segments for the positive and negative halves, with two segments around the zero point forming a single straight line segment. Each segment is divided into linear steps with the segments about the zero point having 32 steps and the rest having 16 steps.

This results in a non-linear quantization of the sampled signal, which has a useful effect on the signal-to-quantizing noise (S/Q) ratio.

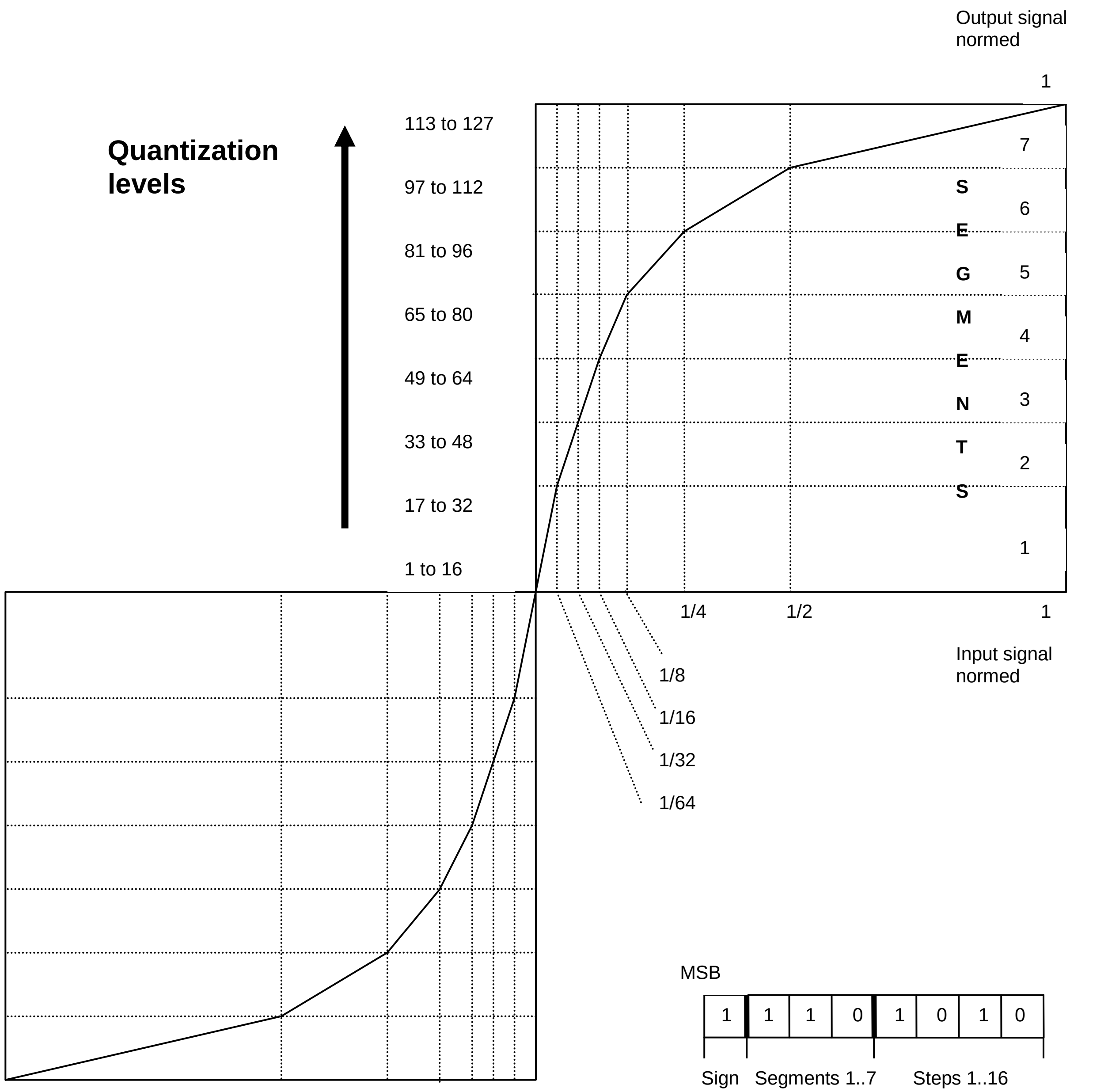
Most of the information in the human voice is at the low amplitudes and the segment about the zero point covers one sixty-fourth of the amplitude range and is divided into 32 steps. Whereas the top half of the input signals dynamic range is covered by the last 16 step segment of the characteristic.

With 128 steps for the positive and negative signal amplitudes a total of 256 steps require 8 bits (2^8). The most significant bit is the sign bit and is set to 1 for the positive amplitudes (0 for the negative amplitudes). The next 3 bits are used for the segments with the last 4 bits for the 16 steps in each segment.



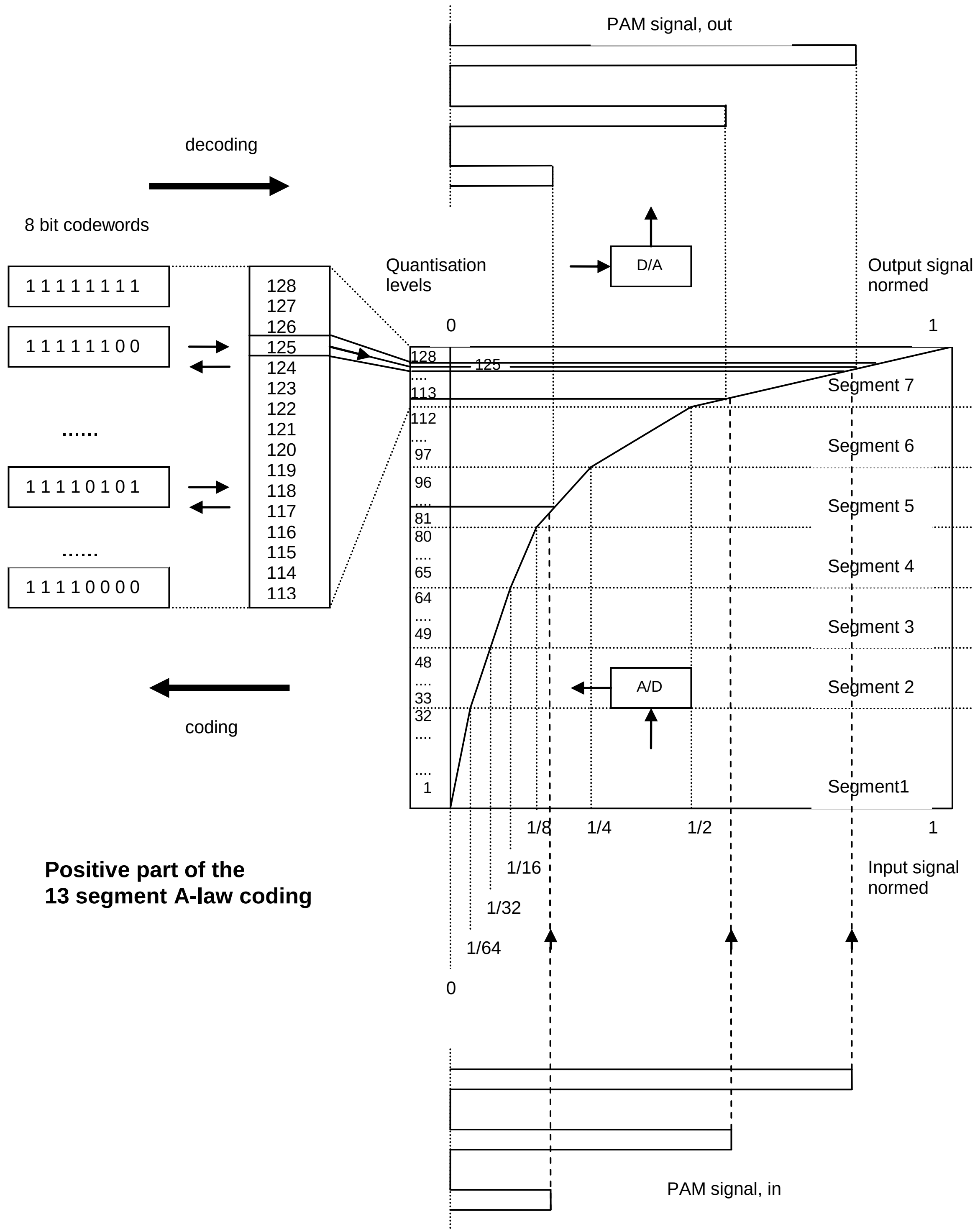
1.3 13-Segment Compression Based upon ITU-T G.711

The following drawing shows the complete 13-segment compression characteristic based upon the ITU-T recommendation G.711



1.3.1 Example of Quantizing a PAM Signal

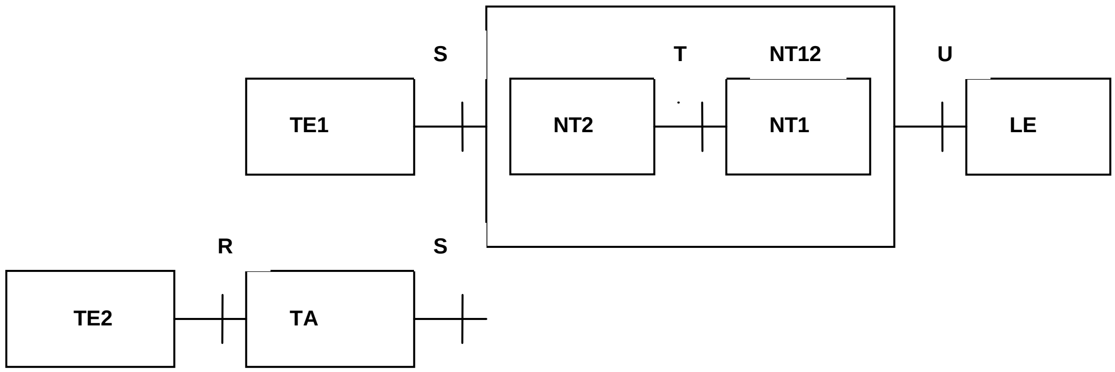
Find below a drawing which shows the quantization process with the A-law coding.



Positive part of the 13 segment A-law coding

1.4 ISDN Functional Devices

For the purpose of ISDN user access, equipment is classified into functional groups. A functional group may consist of one or more items of equipment. The conceptual points dividing the functional groups are described as reference points. The various functional groups and reference points used by ISDN are defined below.



LE ISDN Local Exchange

NT1 Network termination type 1 represents the termination of the physical connection between the customer site and the LE

NT2 Network termination type 2 provides customer site switching, multiplexing and concentration, includes for example PABX. NT2 can be absent

NT12 One device performing NT1 (local loop termination) and NT2 (customer site switching) functions – for example a PABX

TE1 Terminal equipment type 1 are those devices that use the ISDN protocols and support ISDN services, e.g. an ISDN telephone

TE2 Terminal equipment type 2 are non-ISDN devices like e.g. analogue telephones

TA Terminal adaptor allows non-ISDN devices to use the network, converts conventional analogue protocol (a/b) to an ISDN protocol

R Reference point between non-ISDN terminal equipment, TE2, and terminal adaptor, TA.

S Reference point between the ISDN user equipment, TE1 or TA, and the network termination NT1 or NT2.

T Reference point between customer site switching equipment, NT2, and the local loop termination, NT1. In the absence of NT2, the user-network interface is usually called S/T reference point. S and T points are physically identical.

U Reference point between NT1 and LE, also referred to as transmission line.

2. ISDN Network Access

There are two levels of service: the **Basic Rate Interface** (BRI), intended for the home and small enterprise, and the **Primary Rate Interface** (PRI), for larger users. Both rates include a number of B-channels and one D-channel. Each B-channel carries data, voice, and other services. The D-channel carries control and signaling information.

2.1 Basic Rate Interface

The Basic Rate Interface consists of two 64 Kbps (kilobit per second) B-channels and one 16 Kbps D-channel. Thus, a Basic Rate user can have up to 128 Kbps service.

- Access to the network is called Basic Rate Access (BRA).
- It is provided through a Basic Rate Interface (BRI).
- This kind of interface is also called an S_0 Interface.
- There are two channels that you can use.
- The total data rate across this interface 144000 bits per second or 144 Kbps.

This bit rate was chosen because the wiring already installed by the telephone companies under the streets can carry baseband (digital) transmission at this speed.

2.2 Primary Rate Interface

The Primary Rate consists of 30 B-channels - each 64Kbps - in Europe (E1) or 23 B-channels in North-America and Japan (T1) and one 64 Kbps D-channel.

- access to the network is called Primary Rate Access (PRA).
- It is provided through a Primary Rate Interface (PRI).
- This kind of interface is also called an S_2 Interface.
- There are either 30 channels (most of the world) or 23 channels (North America, Japan) that you can use.
- The total data rate across this type of interface is 2048000 bits per second (2Mbits per second) in Europe. In North America and some other countries the total data rate is 1536000 bits per second (1,5Mbits per second)
- This kind of access requires the installation of a high-speed line to the customer premises.

Normally, Basic Rate would be for domestic use, telecommuters or smaller remote offices. Primary Rate would typically be used for large remote access servers, fax servers like KCS or PBXs in medium sized or large offices. For instance, most ISPs (Internet Services Providers) use PRI lines to provide dial-in analogue and ISDN connections for their subscribers.

Note: PBX stands for Private Branch Exchange. PABX means Private Automatic Branch Exchange. Nowadays, these both mean the same thing. This is a telephone exchange as would be used within an office to connect a number of internal users to a usually smaller number of outside lines on the public network.

2.3 ISDN Services

ISDN services are categorized based upon their scope and the source of the service.

2.3.1 Network Services

Network services carry the interactions between the user and the network, for example: setting up calls and disconnecting them. Network Services define how the user and the network interact with each other in order to manage calls. The user can use Network Services to request the network to perform functions such as making and clearing calls, transferring calls to another user, and so on. This activity is known as signaling.

2.3.2 Bearer Services

Bearer services carry data between two users, for example: voice or fax information encoded as a bit stream in real time. These services correspond to the lower three layers of the OSI model. Bearer services carry the call activity that the user is performing at any given moment. This includes voice calls, fax and modem calls, and connections to the Internet. Broadly speaking, there are two forms of bearer service.

- Structured Data - the information passing over the bearer service is in a format that is understood by the network. Voice is an example of structured data. Because the network knows that the connection carrying voice, it can convert the data into an analogue signal in the event that the call is connected to an ordinary analogue phone.

- Unstructured Data - the format of the information is not understood by the network, but is understood by the two users at either end of the service.

Bearer services allow information transfer and involve only lower layer functions (Layer 1). Having chosen the specific bearer service, the users may agree between each other to use any higher layer protocols across the requested connection.

Use of higher layers is transparent to the ISDN and the network makes no effort to assure compatibility between the higher layers. Most important bearer services are:

- Information transfer mode (circuit or packet)
- Information transfer rate (64kbps, 2x64kbps etc. with circuit mode, packet/seconds with packet mode)
- Information transfer capability (Speech, 3,1kHz Audio, 7kHz Audio)

2.3.3 Tele Services

These are value added services provided by the network. Tele services can provide end-to-end communication (user-to-user) and they are characterized by their lower layer attributes (bearer service) and higher layer attributes (value added service). Tele services combine the transportation function with the information-processing functions, which correspond to OSI layers 4 through 7.

What really separates teleservices from bearer services are the higher layer, end-to-end functions. A teleservice may be offered to a user by another user of the network or by the network itself.
Examples of teleservices are telephony, facsimile, videotext and X.400.

2.3.4 Supplementary Services

Supplementary services are used in conjunction with bearer services; they can be seen as features and options.

Supplementary services cover a wide range of capabilities. In general, public ISDN providers will offer them on an extra-change basis, by subscription or per use.

Examples of supplementary services are direct dialing in (DDI), multiple subscriber number (MSN), call waiting, advice of charge (AOC), etc..

3. ISDN B and D Channels

The Basic Rate Interface consists of two 64 Kbps B-channels and one 16 Kbps D-channel. The Primary Rate consists of 30 B-channels in Europe (E1) or 23 B-channels in North-America and Japan (T1) and one 64 Kpbs D-channel.

3.1 The B-Channel

In the ISDN network, the B-channel is the channel that carries the main data. The "B" stands for "bearer" channel.

The B channel carries ISDN Bearer Services across the network and so carries the content of call (the voice, fax or data) between users. It carries data at 64000 bits per second (56000 bits per second in some North American networks).

The ISDN does not need to know what the bits represent. The job of the network is to accept a stream of bits supplied by one user at one end of the B channel and to deliver them to the other user at the opposite end of the channel.

Within an interface, the B channels are numbered. In a Basic Rate Interface they are numbered 1 & 2; in a Primary Rate Interface, they are numbered 1 to 30 (or 23 in North America and Japan). When two users are connected, there is no relationship between the channel numbers used at each end. You might have one user's B channel number 17 connected with the other user's B channel number 2. The ISDN is responsible for managing this relationship.

Notice that channel number 17 would only be possible on a PRI, while channel number 2 is possible on both a BRI and a PRI. ISDN does not restrict the interconnection of B channels between the two kinds of interface.

3.2 The D-Channel

In the ISDN network, the D-channel is the channel that carries control and signaling information. The "D" stands for "delta" channel.

The D channel carries the ISDN Network Services between the user and the network. It maintains the user's relationship with the network.

This includes:

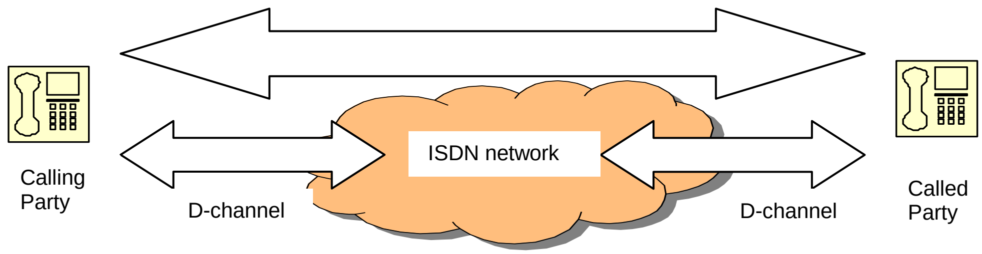
- the requests and responses used when you make or receive a call
- call progress messages
- messages informing you that the called party has closed the call
- error messages telling you why a call has not been established for you

The D channel operates at 16000 bits per second (16kbps) in a BRI and at 64000 bits per second (64kbps) in a PRI.

3.3 B and D Channel Characteristics

An ISDN channel has two and only two ends. B channels terminate at a user. A B channel can therefore connect two and only two users. A B channel cannot be Y-shaped. B channels are therefore described as end-to-end.

In the case of the D channel one end is with the user. The other end is in the network. A D channel is not end-to-end. You cannot normally, therefore, use a D channel to carry data between two users.



Notice how the D channels in the picture above do not pass through the network. Notice also how each user has only one D channel and it is not connected in any way with the D channel of the other user. The B channel passes directly across the network.

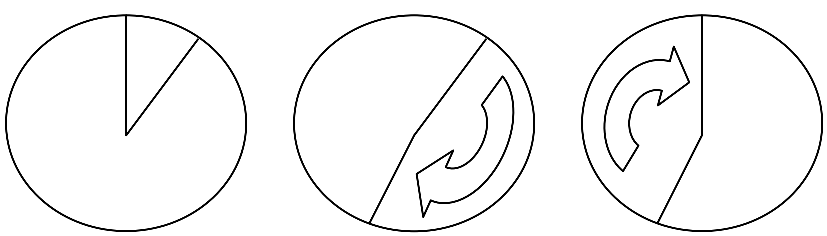
3.4 B and D Channel Line Sharing for Basic Rate

The two B channels and the one D channel that make up a Basic Rate ISDN line are assembled together within the interface using a technique called Time Division Multiplexing (TDM).

TDM is a scheme in which numerous signals are combined for transmission on a single communications line or channel. Each signal is broken up into many segments, each having very short duration.

The circuit that combines signals at the source (transmitting) end of a communications link is known as a multiplexer. It accepts the input from each individual end user, breaks each signal into segments, and assigns the segments to the composite signal in a rotating, repeating sequence. The composite signal thus contains data from all the end users. At the other end of the long-distance cable, the individual signals are separated out by means of a circuit called a demultiplexer, and routed to the proper end users. A two-way communications circuit requires a multiplexer/demultiplexer at each end of the long-distance, high-bandwidth cable.

It works like this. Imagine a clock with the second hand spinning. For a portion of the arc described by the second hand, the interface is carrying data for the D channel; for another portion of the arc, the interface is carrying data for a B channel, see the drawing below.



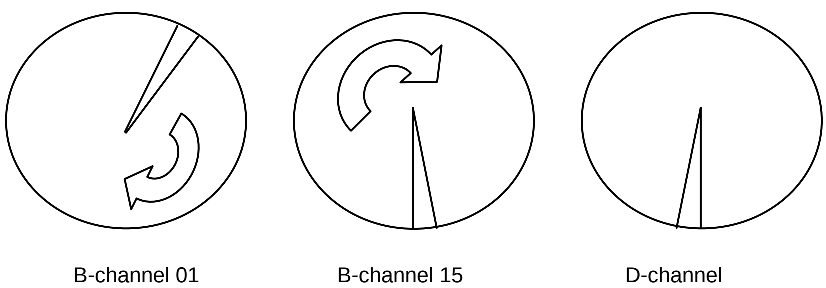
D-channel First B- channel Second B- channel

3.5 B and D Channel Line Sharing for Primary Rate

A EuroISDN Primary Rate Interface contains 30 B Channels and one D channel. In North America and some other countries a PRI contains 23 B channels and one D channel.

A B channel operates at 64000 bits per second. (56000 bits per second in some parts of North America and other countries.)

A D channel operates at 64000 bits per second in a Primary Rate Interface. Note that this is different to the BRI where the D channel runs at 16000 bits per second.

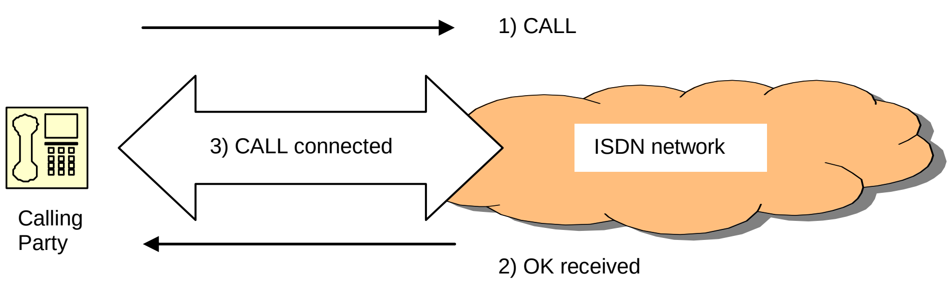


The Primary Rate Interface spends equal amounts of time transmitting data for each of the B channels and for the D channel because they all operate at the same speed. The D channel appears between B channel 15 and B channel 16.

3.6 Fractional B-Channels for Primary Rate

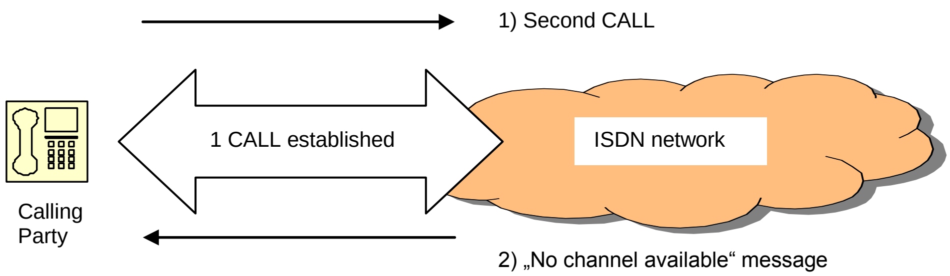
ISDN service suppliers and PABX systems have the opportunity to supply interfaces where not all the channels are active. In most countries, when you take a Primary Rate Interface, you are charged a rental per channel. If you don't need all the channels that are available, you can ask to have some of these channels deactivated. This is known as **fractional Primary Rate**.

What happens if you try to use more channels than you actually are connected to? In ISDN, the network is the arbitrator of everything. When you want to place a call through an ISDN network, you send a request in your D channel to the network. The network will either attempt to satisfy your request or it will refuse it with an error “no channel available”.



In the example above the calling party initiates a CALL and receives an OK from the network. After the OK has been received, the CALL is connected to its destination and one B-channel is in use.

If the Calling party is using a fractional ISDN connection with just one B-channel available, the following happens if a second CALL is initiated while the first CALL is still established.



3.7 B and D Channel Protocol

You must use a protocol to establish meaningful communication across a channel. It is important that both parties (calling and called party) use the same protocol.

This is particularly important for the D channel. Your signaling requests and responses must be understandable by the network. Even if your ISDN device and ISDN line are both functioning correctly, you might not be able to make successful calls if you're using a D channel protocol that isn't the same as the network's.

The most general international ISDN protocol is the **Q.931** defined by the ITU-T for signaling in the D channel. But this protocol is not really being used anywhere in the world. However, there are several signaling protocols use in the public networks based on (or derived from) Q.931 in use round the world: For instance, AT&T 4ESS and AT&T 5ESS are used in North America while much of the rest of the world is now using EuroISDN (also called ETSI or DSS1). 1TR6 is a protocol specially used in Germany but it’s now more and more replaced by the EuroISDN.
On the other hand, the QSIG protocol is being used in private networks for inter-exchange signaling between two PABX systems.

The supported ISDN protocols for KCS are DSS1 (EuroISDN), 1TR6, QSIG, 4ESS.

3.7.1 Service Identification with Protocol 1TR6

With the 1TR6 protocol, an outgoing call must contain a service identification also called SI. This SI can be used by the receiver, or the PABX, as information whether the call should be answered or not. This feature can be used to use the same number for different devices.

The SI used by KCS can be configured via the ISDN configuration, line number 251. The following values are possible:

- Value 11, Service group telephone, service ISDN telephone 3,1kHz
- Value 12, Service group telephone, service analogue telephone
- Value 22, Service group a/b services, service fax group 3

The default value for the service identification is defined as 00 which will be interpreted as 22, fax group3. Since KCS is a fax group 3 machine, this value should be the appropriate setting. Practical tests showed that not every wanted destination can be reached with that identification (digital telephone, fax machines via wrong configured PABX systems, terminal adapters). In that case it is suggested to use 11 as service identification, as the service “ISDN telephone 3,1kHz” will get a connection to both, digital faxes and analogue lines.

3.8 B-Channel Characteristics

It is important to remember that ISDN channels cannot be divided up into smaller units. Each is provided on an "all or nothing" basis.

Two users communicating over a B channel have 64000 bits per second available to them. There is nothing they can do to reduce this bandwidth.

What about the situation where the two users find that 64000 bits per second is not sufficient? The only solution is to add another B channel. This gives them 128000 bits per second. They are not using a single B channel of 128000 bits per second. (Don't forget that the speed of a B channel is defined as 64000 bits per second. Anything which operates at a different speed isn't a B channel.)

This means that they will have two parallel calls between them and the phone bill will show two simultaneous calls.

4. ISDN PROTOCOL ARCHITECTURE

ITU-T ISDN Protocols are specified only across the S/T reference points and only on the D-channel. They are roughly equivalent to the lower three layers of the OSI Reference Model. Since these protocols describe only the user-network interface and no user-to-user communication, there are no D-channel counterparts for the OSI end-to-end layers.

On the other hand, the user may choose any protocol(s) for the B-channels (for example fax).

4.1 ISDN Layer 1 Definition

Layer 1 is called “Physical Layer” and provides the electrical, mechanical, functional and procedural characteristics to activate, maintain and deactivate the physical connection

4.1.1 ISDN Basic Rate Layer 1

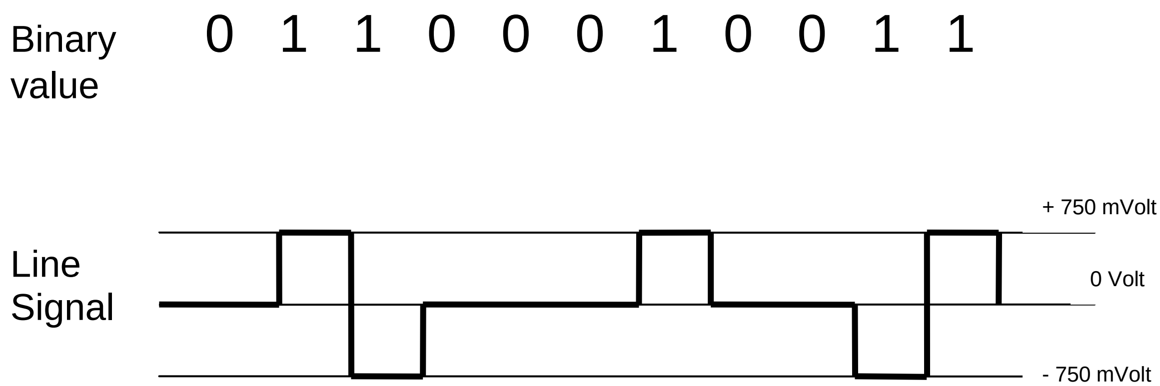
The Terminal equipment derives its timing (bit, octet and frame) from the signal received from the NT and uses this derived timing to synchronize its transmitted signal. The nominal bit rate is 192kbit/s. This bit rate is defined as follows:

The Basic Rate ISDN Interface structure is composed of two 64kbit/s B channels and one 16kbit/s D channels, and hence is generally referred to as ‘2B + D’. Thus the Basic Rate Interface Structure has a rate of 2x 64kbit/s + 1x16kbit/s in sum 144 kbit/s. Additionally, the S/T interfaces require an extra 48 kbit/s for synchronization and maintenance and thus has a working rate of 192kbit/s.

4.1.1.1 ISDN Basic Rate Line Coding

Alternate Mark Inversion (AMI) line coding is used in which binary zeros are transmitted as no line signal (space) and binary ones are transmitted as either a positive or negative pulse (mark). Additionally **Pseudoternary coding**, a variation of AMI with logical 0´s transmitted as marks and logical 1´s transmitted as spaces is used for Basic rate interface at reference points S/T. The polarity of subsequent binary ones must alternate. The nominal pulse width is 5,21 µsec (100% pulse widths used) and the nominal pulse amplitude is 750mV (zero to peak). The following drawing shows how this is working:

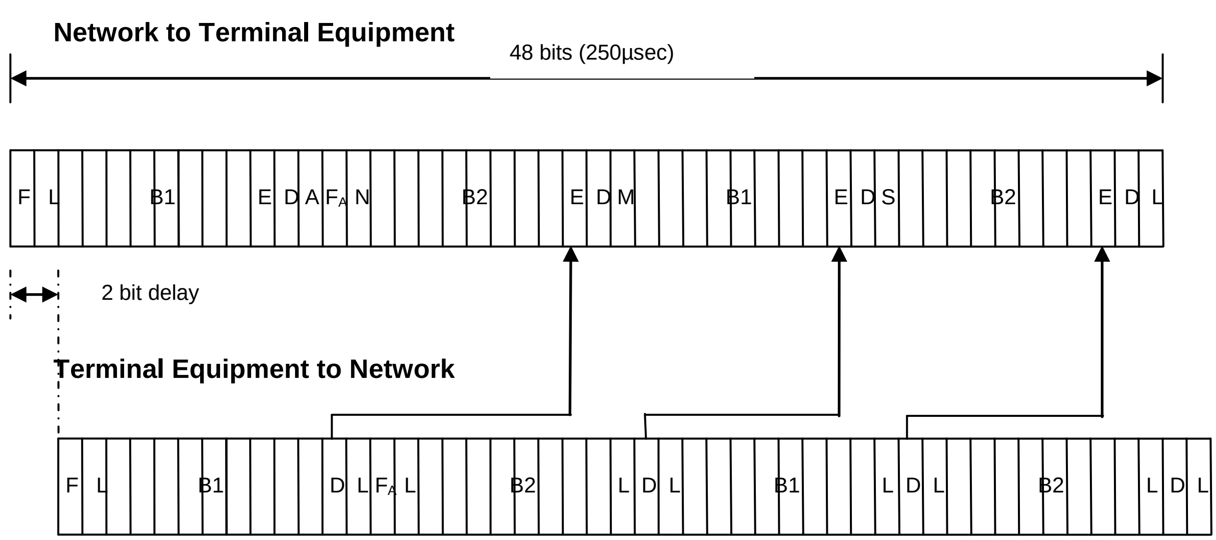
Pulse width = 1/bitrate or 1/192000 = 0,000005208 = 5,21 µsec



4.1.1.2 ISDN Basic Rate Line Framing

As with all TDM (Time Division Multiplexing) schemes, the 2 B + D channels are multiplexed together along with 48kbit/s capacity for framing and synchronizing purposes, into repetitive fixed length frames. The Layer 1 Frame has a length of 48 bits. Frame structures differ slightly for each direction of transmission. The frame structures for both directions are illustrated below.

There is a 2 bit delay between the start of the frame received from the NT by the Terminal Equipment and start of the frame transmitted by the Terminal Equipment to the NT. (The corresponding offset at the network will be greater due to delays caused by the interface cable.) The information contained in the D channel is echoed back to the TE by the NT for the purpose of access control. The delay between transmission of the D bit and receipt of the echoed bit is either 8 or 10 bits depending on the position within the frame.



F ….	Framing bit	N ….	Bit set to FA
L ….	DC balancing bit	B1 ...	B-channel 1
D ….	D-channel bit	B2 …	B-channel 2
E ….	D-echo channel bit	A ….	Bit used for activation
FA …	Auxiliary framing bit	S ….	Use of this bit is for future study
		M ….	Multiframing bit

The beginning of the frame is marked by the **F** (framing) bit. To avoid polarization, the **L** (DC balancing) bit is also transmitted to balance the polarity of this framing bit. The **F** bit is always binary zero and always of the same polarity as the preceding pulse and thus a line code violation. Use of a line code violation to mark the start of the frame allows rapid reframing.

According to the coding rule, both the **F** bit and the first binary zero following the **F/L** pair produce a line code violation. Consequently the $\mathbf{F_A}$ **/N** pair (NT to TE) or $\mathbf{F_A}$**/L** pair (TE to NT) are used to ensure that there is a line code violation at 14 bits or less from the framing bit **F**.

The **E** (D echo channel) bit is used to echo back **D** channel bits arriving at the NT. Echoing of the D channel is used for D channel access control. The **A** (activation) bit is used for the activation procedure and indicates to terminals that synchronization is in place. The **M** (multi-framing) bit is to be used in conjunction with an additional channel of traffic, the Q channel. The Q channel is set by ensuring the **M** bit is binary one on every twentieth frame and the $\mathbf{F_A}$ bit is binary one on every fifth frame (i.e. in each 2Q frame multi-frame there are 4Q bits). The last **L** bit of the frame is used for balancing the whole frame.

4.1.2 ISDN Primary Rate E1 Layer 1

The Terminal equipment derives its timing (bit, octet and frame) from the signal received from the NT and uses this derived timing to synchronize its transmitted signal. The nominal bit rate is 2048kbit/s. This bit rate is defined as follows:

The Primary Rate ISDN Interface structure is composed of thirty 64kbit/s B channels, one 64kbit/s D channel and one 64kbit/s channel which is used for a specific frame alignment signal (FAS). Thus the Primary Rate Interface Structure has a rate of 32 x 64kbit/s in sum 2048kbit/s or 2,048Mbit/s.

4.1.2.1 ISDN Primary Rate E1 Line Coding

Primary Rate ISDN codes bits using High-Density Bipolar 3 (HDB3) coding which utilizes Alternate Mark Inversion (AMI) coding rules to achieve DC balance of the transmitted signal. However, HDB3 applies additional coding rules to ensure that the timing signal can still be retrieved when **long sequences of consecutive logical zeros** are transmitted.

The rules for HDB3 coding are described below:

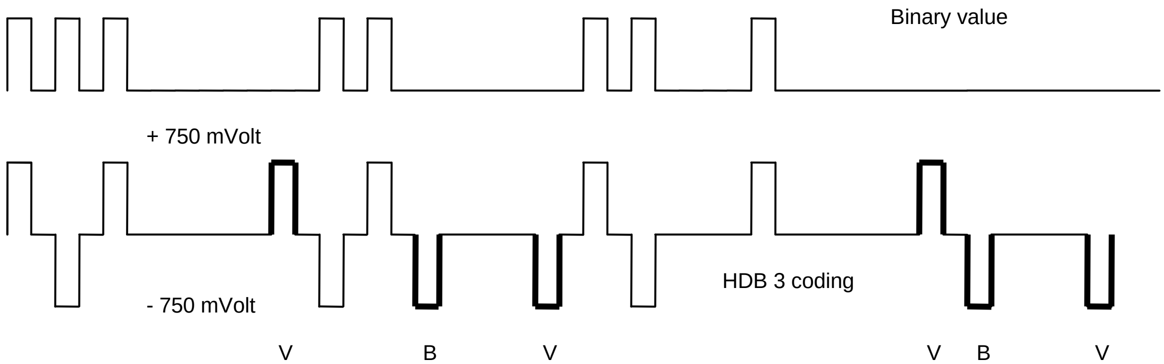
1. The binary bits are encoded using AMI coding where binary 0s are transmitted as a space (no pulse) and binary 1s are transmitted as a pulse (mark) of which the polarity is alternated.
2. Because timing information can only be extracted where pulses are present, where strings of consecutive spaces occur timing information cannot be obtained. HDB3 never allows more than three spaces to be transmitted consecutively.

   Thus, where ever four binary 0s occur consecutively within the data stream, the forth binary 0 is transmitted as a mark rather than a space. To indicate that the mark relates to a 0 rather than a 1, the mark violates AMI coding rules by being transmitted with the same polarity as the previously transmitted mark. This binary 0 transmitted as a mark is hence referred to as a violation bit or V bit.

3. Because V bits violate AMI coding rules, they could cause DC balance to be lost. Hence whenever there is an even number of marks between violation bits (i.e. both violation bits have same polarity), the first of the four zeros is also transmitted as a mark rather than a space, but without violating AMI rules. This will cause the V bit to be transmitted with opposite polarity to the previous V bit and hence DC balance will be maintained. Consequently, this first binary 0 bit if transferred as a mark is referred to as the Balance bit or B bit.

An example of an HDB3 encoded data stream is shown in the drawing below:

1 1 1 0 0 0 0 1 1 0 0 0 0 1 1 0 0 1 0 0 0 0 0 0 0 0

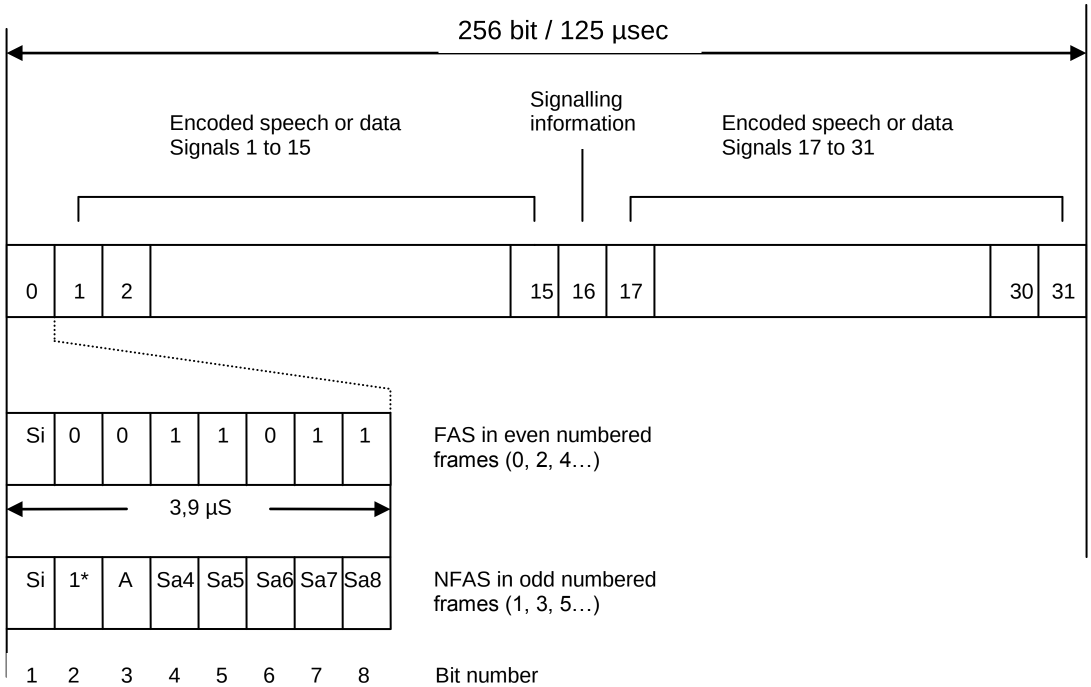


V …. Violation bit
B …. Balance bit

Because the first V is of the same polarity as the previous 1, this is an obvious violation of the AMI rules, therefore the receiving equipment knows that the V bit represents another zero. Likewise where a B and V bit have been inserted, together they have violated the AMI rules, thus they represent 4 consecutive zeros.

4.1.2.2 ISDN Primary Rate E1 Line Framing

The standard frame structure for Primary rate ISDN consists of 32 timeslots numbered from 0 to 31. 30 timeslots (number 1 to 15 and 17 to 31) are assigned to voice/data channels or also called B-channels; timeslot 16 is assigned for signaling information also called the D-channel. Timeslot 0 is assigned for the Frame Alignment Signal (FAS) and the Non-Frame Alignment Signal (NFAS) which are transmitted in alternate frames.



Si …. Bits reserved for international use
A …. Remote alarm indication. Set to 1 in alarm condition. Sa4 to Sa8 are spare bits
1* …. This bit is fixed at 1 to assist in synchronizing the frames

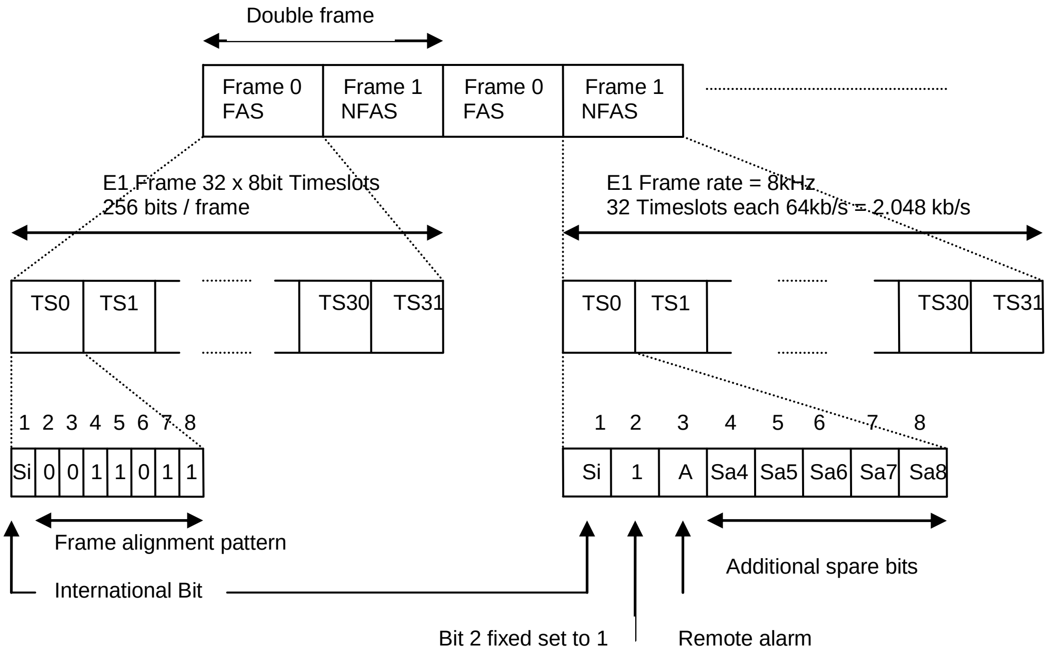
The basic frame format is combined into various E1 multiframe formats, e.g FAS/NFAS doubleframe framing or CRC-4 multiframing which are the two common framing methods for E1 connections.

FAS/NFAS doubleframe framing, two distinct basic frame types are defined, one with the Frame Alignment Signal (FAS) word in timeslot TS0, and one with the Non-Frame Alignment Signal (NFAS) word in timeslot TS0. These two different frame types are alternatively transmitted, and are used to determine frame synchronization, and provide bandwidth for maintenance and overhead functions.

In order to achieve synchronization, the following detection sequence is required:

1. Searching for the FAS word - Si 0 0 1 1 0 1 1.
2. Then checking the next frame for the NFAS bit 2 being set to 1 - Si 1 A Sa4 Sa5 Sa6 Sa7 Sa8.
3. Then checking again for the FAS - Si 0 0 1 1 0 1 1 in the following frame period.

With these 3 conditions being satisfied then synchronization is achieved. Once frame synchronization has been achieved, the multiplexer can separate the individual channels in the frame. If 3 out of 4 Frame Alignment Signals are received in error, then synchronization is deemed to be lost and the terminal commences the synchronization procedure.



As an alternative and usually the preferred method, **CRC-4 multiframe** framing (four-bit cyclic redundancy check) is used for immunity against false framing and also provides non-intrusive error monitoring capabilities for the E1 payload data. Implementation consists of redefinition of Bit 1 of the FAS/NFAS frames, and definition of a larger multiframe structure.

The CRC-4 multiframe is comprised of 16 alternating FAS/NFAS frames, consecutively numbered from 0 to 15. This multiframe is in turn divided into two 8-frame “sub-multiframes”, known as SMF I and SMF II. When CRC-4 multiframing is enabled, a CRC is calculated for each “sub-multiframe” and is reported in the next multiframe.

In the frames containing FAS, Bit 1 is used to send the four CRC-4 bits, designated C1 – C4, in each SMF. In NFAS frames, Bit 1 is used to transmit the six bit CRC-4 multiframe alignment pattern 001011 and two CRC-4 error indication bits (E).

The following table shows a general overview about the CRC-4 multiframe structure as described previously:

SMF	Frame number			Timeslot 0 Bit number					
		1	2	3	4	5	6	7	8
	0	C1	0	0	1	1	0	1	1
	1	0	1	A	Sa4	Sa5	Sa6	Sa7	Sa8
	2	C2	0	0	1	1	0	1	1
	3	0	1	A	Sa4	Sa5	Sa6	Sa7	Sa8
	4	C3	0	0	1	1	0	1	1
	5	1	1	A	Sa4	Sa5	Sa6	Sa7	Sa8
	6	C4	0	0	1	1	0	1	1
	7	0	1	A	Sa4	Sa5	Sa6	Sa7	Sa8
	8	C1	0	0	1	1	0	1	1
	9	1	1	A	Sa4	Sa5	Sa6	Sa7	Sa8

10	C2	0	0	1	1	0	1	1
11	1	1	A	Sa4	Sa5	Sa6	Sa7	Sa8
12	C3	0	0	1	1	0	1	1
13	E1	1	A	Sa4	Sa5	Sa6	Sa7	Sa8
14	C4	0	0	1	1	0	1	1
15	E2	1	A	Sa4	Sa5	Sa6	Sa7	Sa8

C1 – C4	CRC-4 bits
E1 – E2	Error indication bits
A	remote alarm indication
Sa4 – Sa8	additional spare bits

4.1.3 ISDN Primary Rate T1 Layer 1

The Terminal equipment derives its timing (bit, octet and frame) from the signal received from the NT and uses this derived timing to synchronize its transmitted signal. The nominal bit rate is 1544kbit/s. This bit rate is defined as follows:

The Primary Rate ISDN Interface structure is composed of twenty three 64kbit/s B channels and one 64kbit/s D channel. One PRI frame contains 1 framing bit (F) plus a single 8-bit sample from each of the 24 channels. Thus the Primary Rate Interface Structure has a rate of 8 bit x 24 channels +1 framing bit in sum 193 bits per frame. With 8000 frames per second, the total bit rate is 8000 x 193 = 1544kbit/s. From this 1544kbit/s is 1536kbit/s available as user data.

4.1.3.1 ISDN T1 Line Coding

T1 Primary Rate ISDN codes bits using the AMI with ZCS coding (alternate mark inversion with zero code substitution) or B8ZS (Bipolar 8 Zero Substitution) coding rules to achieve DC balance of the transmitted signal

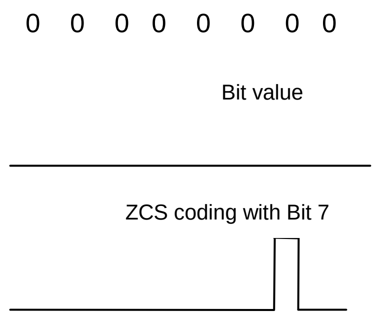
The AMI (ZCS) coding is already described in a previous chapter and is the old method. The preferred method for T1 line coding is the B8ZS coding which has been implemented to replace AMI (ZCS).

The general two rules for ones density for T1 connections are defined as follows:

1. There can be no more than 15 consecutive zeros in a T1 stream.
2. The average ones density must be at least 12,5 percent (one “1” in 8 zeros on average).

Some network clocking devices may lose synchronization after having received as few as eight consecutive zeros. Therefore, it is necessary (an AT&T rule) to maintain a certain “ones density.”

The first method, ZCS (zero code substitution) is described in the drawing below:

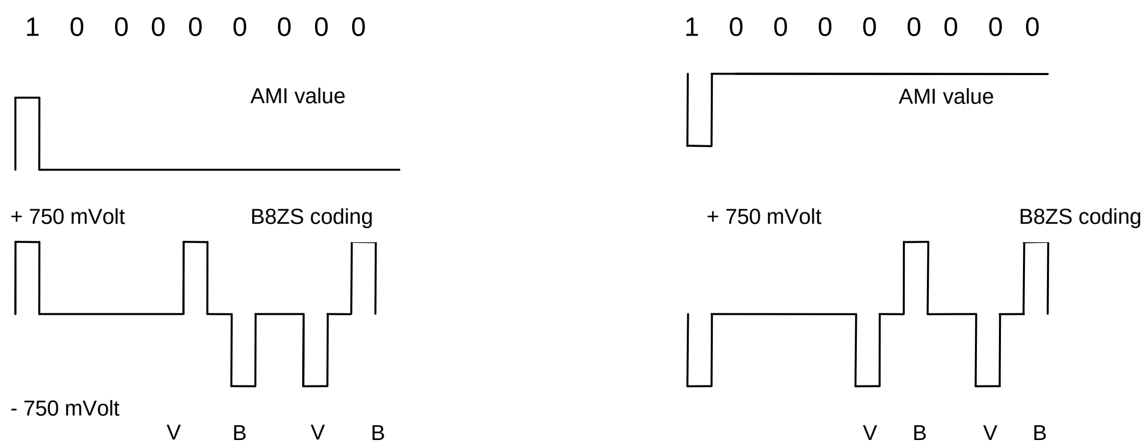


This method is also called “jam bit 7” or “bit 7 stealing” or “bit robbing”. Bit seven was chosen to avoid destroying the signaling or control bits carried in bit position eight.

The most common method of ensuring Ones Density is “Zero Code Suppression” (ZCS). ZCS ensures One's Density by setting one bit of each 8 bit word to a one. This effectively makes only seven bits available for data. Seven bits per sample times 8000 samples per second equals 56000 bits per second available for data. So this method has the disadvantage that instead of 64kbit/s only 56kbit/s are available.

Therefore a new coding has been implemented which replaces this ZCS, called B8ZS.

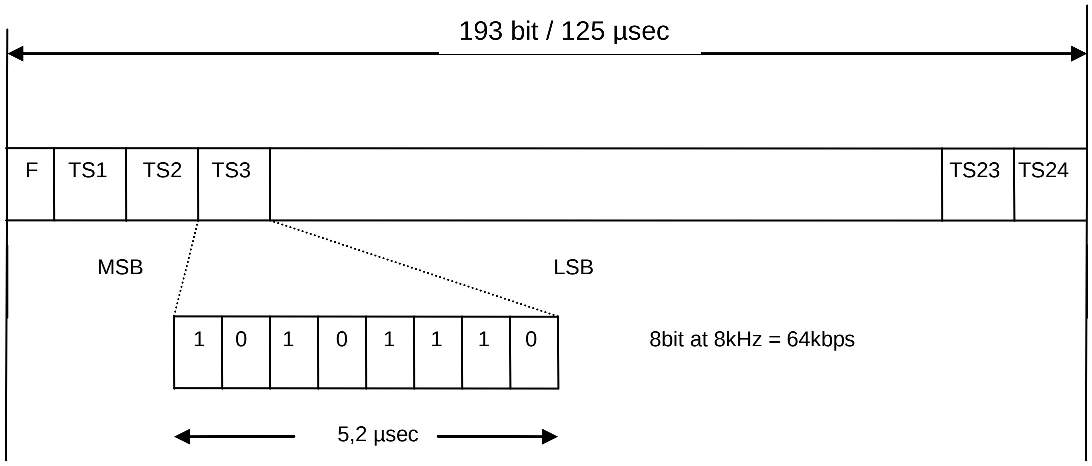
With B8ZS a logical "0" is represented by the absence of a line signal and a logical "1" by alternating positive and negative impulse, just the opposite of the BRI-interface encoding scheme. In the B8ZS-scheme there is also a measure provided for the case that a string of no line signals appear. If 8 consecutive zeros in a row appear this series is substituted by the following series: the first three are unchanged, violation, correct 1, 0, violation, correct 1.



V …. Violation bit
B …. Balance bit or correct bit

4.1.3.2 ISDN T1 Line Framing

The basic T1 frame begins with 1 framing bit F, followed by 192 payload bits. The payload is divided into 24 Timeslots TS1…TS24 consisting of 8 bits per Timeslot. The bit rate for the T1 frame is fixed at 1544 kbit/s. The standard frame structure of a T1 primary rate ISDN connection is shown in the picture below:



MSB …. Most significant bit
LSB …. Least significant bit
TSn …. Timeslot 1…24
F …. Single framing bit

The basic frame format is combined into various T1 multiframe formats, e.g Superframe (SF)/ D4 framing or extended superframe (ESF) which are the two common framing methods for T1 connections.

The **T1 Superframe format (also called D4 framing)**, consists of 12 consecutive basic T1 frames, with 1 F-bit and 192 payload bits per frame (in sum 2316 bits). The 12 F-bits are divided into two groups, described as follows:

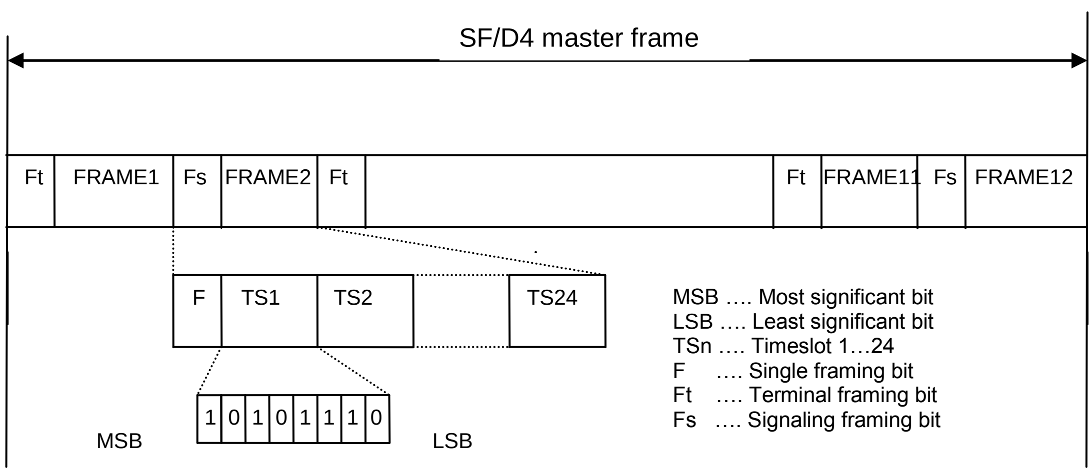
- Six terminal framing (Ft) bits, used to identify frame boundaries
- Six signaling framing (Fs) bits, used to identify robbed-bit signaling frames and superframe boundaries.

The receiver detects SF/D4 frame synchronization when a pre-determined number of consecutive correct framing bits (Ft and Fs) are detected.

F - bits

Frame number	Bit number	Terminal framing bit Ft	Signaling framing bit Fs
1	1	1	-
2	194	-	0
3	387	0	-
4	580	-	0
5	773	1	-
6	966	-	1
7	1159	0	-
8	1352	-	1
9	1545	1	-
10	1738	-	1
11	1931	0	-
12	2124	-	0

The drawing below shows a complete setup of such a frame



The T1 **extended Superframe (ESF)** format consists of 24 consecutive basic T1 frames with 1 F-bit and 192 payload bits per frame (in sum 4632 bits). The corresponding 24 F-bits are used for a variety of functions, described as follows

- **Framing** (a 2kbit pattern) – This is the terminal synchronization channel where frame and superframe alignment is provided by the F-bit of frames 4, 8, 12, 16, 20 and 24. This sequence is referred to as the Framing Pattern Sequence (FPS). Bits that comprise this sequence are referred to as Fe-bits. The repeating pattern is 001011 binary (uses 6 F-bits per extended superframe).
- **Error detection** (a 2kbit/s pattern) – The cyclic redundancy check (CRC bits) carrying the CRC-6 code of the preceding superframe is located in F-bit of frames 2,6,10,14,18,22 (uses 6 F-bits per extended superframe).

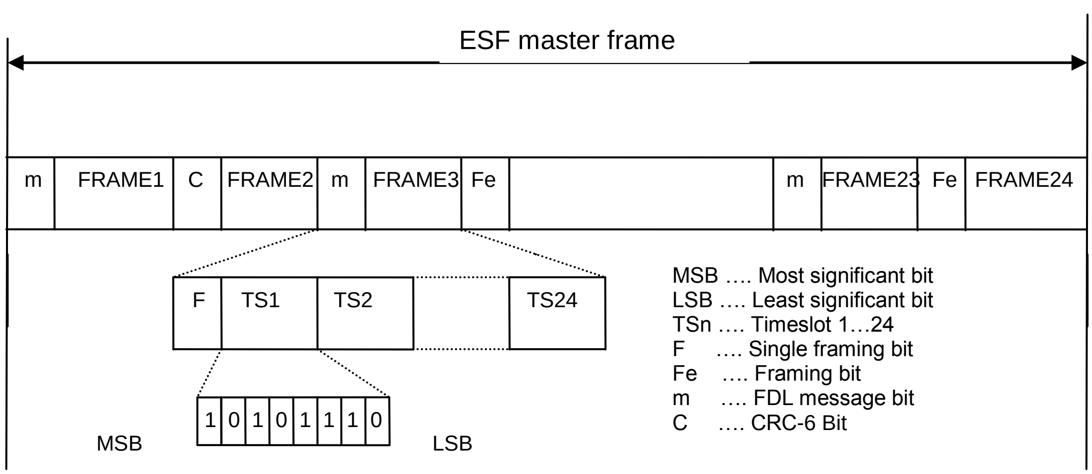
- **Facility Data Link** (FDL bits, a 4kbit/s pattern) – carried by the odd F-bits using 12 F-bits per extended superframe.

Multiframe alignment in ESF mode requires that the proper ordering of Fe bits be found before multiframe alignment is declared.

F - bits

Frame number	Bit number	Framing bit Fe	Facility Data Link	CRC-6
1	1	-	m	-
2	194	-	-	e1
3	387	-	m	-
4	580	0	-	-
5	773	-	m	-
6	966	-	-	e2
7	1159	-	m	-
8	1352	0	-	-
9	1545	-	m	-
10	1738	-	-	e3
11	1931	-	m	-
12	2124	1	-	-
13	2317	-	m	-
14	2510	-	-	e4
15	2703	-	m	-
16	2896	0	-	-
17	3089	-	m	-
18	3282	-	-	e5
19	3475	-	m	-
20	3668	1	-	-
21	3861	-	m	-
22	4054	-	-	e6
23	4247	-	m	-
24	4440	1	-	-

The drawing below shows a complete setup of such a frame



On KCS, the mode of operation for the primary rate Layer 1 is defined within ISDN config line 291, second position.

:00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,291

↓

Position 2, Primary rate Layer 1 operation mode for Europe E1 and USA/Japan T1

HEX 00 in Binary writing: 0 0 0 0 0 0 0 0
Bit 8 …. Bit 1

Bit 1, value 1	00 …. E1 CRC-4 frame or T1 ESF - extended superframe
Bit 1, value 1	01 …. E1 FAS/NFAS double frame or T1 SF – superframe D4
Bit 2, value 2	00 …. T1 only, enable performance report message, PRM, according AT&T TR54016
Bit 2, value 2	01 …. T1 only, disable performance report message, PRM, according AT&T TR54016
Bit 3, value 4	00 …. T1 only, disable one-second PRM´s according ANSI T1.403
Bit 3, value 4	01 …. T1 only, enable one-second PRM´s according ANSI T1.403
Bit 4, value 8	00 …. T1 only, B8ZS line coding
Bit 4, value 8	01 …. T1 only, AMI with ZCS line coding

Used abbreviations:

CRC-4	E1 cyclical redundancy check 4 multiframe structure
B8ZS	T1 Bipolar with 8 Zeros Substitution
AMI	Alternate mark inversion
ZCS	T1 Zero Code Suppression

Please note: You have to add the bitvalues according to the used coding and framing. The bitvalue for Bit 1 is 1, for Bit 2 is 2, for Bit 3 is 4 and for Bit 4 is 8. So by using T1 Superframe with ZCS line coding and one second PRM´s, the bitvalues for Bit 1, Bit 3 and Bit 4, 1 + 4 + 8 must be added. The resulting value, 13, must be defined as HEX value within config position 2 of line 291. In this case 0D has to be defined!

4.1.4 Summary of Differences Between E1 and T1

The following table shows a summary of the main differences between a T1 and E1 primary rate ISDN installation:

T1 / E1 comparison

Topics	T1	E1
Data rate	1544kb/s	2048kb/s
Effective throughput	1536kb/s	1920kb/s
Number of time slots	24	32
Usable time slots	24	30
Line coding	ZCS, B8ZS	AMI with HDB3
Framing formats	SF or D4, ESF	Double frame, CRC-4
Type of PCM encoding	µ-Law	A-Law
Voice signaling	Robbed bit signaling	Channel associated Signaling (CAS) or Common channel Signaling (CCS

4.2 ISDN Layer 2 Definition

Once Layer 1 separates the D-channel from the B-channels, the D-channel needs a Layer 2 protocol to support its call control and packet handling functions. ISDN Layer 2 is concerned with conveying the layer 3 information across the ISDN interface using the D Channel of the layer 1 frames. The defined ISDN Layer 2 protocol is LAPD

Link

Access

Procedure for the

D Channel

The functions specified by LAPD include:

- Provision of one or more data link connections on a D Channel
- Frame structuring
- Sequence control
- Detection and recovery from transmission, format and operational errors
- Notification to management entity of unrecoverable errors
- Flow control

4.2.1 Layer 2 Differences Between Primary and Basic Rate ISDN

The ISDN primary rate layer 2 protocol is very similar to the ISDN basic rate layer 2 protocol. The differences between the two are summarized below:

- Primary rate provides 30 B channels (or 23 B channels) and one 64kbit/s D channel, whereas Basic rate provides 2 B channels and one 16kbit/s D channel
- Primary Rate is point to point, where as Basic Rate can be point to multipoint;
- Primary Rate normally uses a fixed TEI (Terminal endpoint identifier) value of ‘0’, hence TEI assignment procedures are not normally employed. Primary Rate does still use the ID check procedure (with the purpose of verifying operation - rather than checking a TEI value)

As there is no other difference between basic and primary rate ISDN, the following description is valid for both types of connection.

4.2.2 Layer 2 Frames

Data is transferred across the ISDN data link (in the D channel) in units known as F r a m e s. As well as containing the layer 3 information, the layer 2 frames also contain addressing, control and error checking information applicable solely to transmission across the data link.

Three categories of layer 2 Frame are used:

- **Unnumbered** (SABME – Set Asynchronous Balanced Mode Extended, UI – Unnumbered information, UA -Unnumbered Acknowledgement, DISC – DISConnect, DM - Disconnected Mode, FRMR -Frame Reject
- **Information** (I – Information)
- **Supervisory** (RR - Receiver Ready, RNR - Receiver Not Ready, REJ – Reject)

Unnumbered frames (U Frames) are used for data link establishment and control and for unacknowledged information transfer.

Information frames (I Frames) are sequenced frames used to carry the layer 3 data, once the data link is established.

Supervisory frames (S Frames) are used for flow control, recovery and acknowledgment of I Frames whilst the data link is established. These types of frames contain purely layer 2 information.

All Layer 2 frames are considered as either a Command or a Response frame. Command frames are used to transfer information to or request information from the other side. Response frames are normally used to respond to command frames. (In certain cases response frames may be used which are not in response to a command, these are referred to as unsolicited responses).

Information and Unnumbered frames have defined command/response roles, i.e. the various frame types are used exclusively as commands or exclusively as responses. Supervisory frames are normally used as commands but can also be used as responses when required.

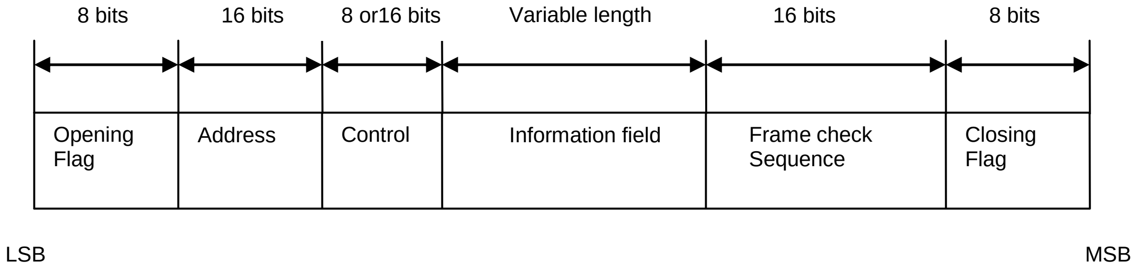
4.2.3 Layer 2 Frame Structure

The Layer 2 frames consist of a number of fields of data, as follows:

- Address Field
- Control Field
- Information Field (not present in all frame types)
- Frame Check Sequence

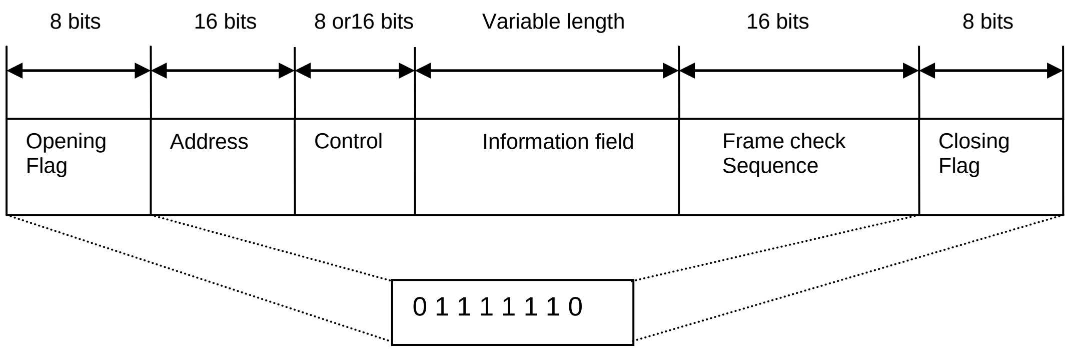
In addition, to indicate the start and end of a frame, the frame is preceded and ended with a flag (a defined reserved data pattern). The ISDN Layer 2 frame structure is shown below. Each of the various fields are described in further detail below.

Layer 2 Frame check sequence



All frames start and end with a flag, see below. The one preceding the address field is defined as the ‘opening flag’ and the one following the FCS field is defined as the ‘closing flag’. All flag sequences start with 0 followed by six continuous 1’s and a 0 again, i.e. Flag is always 01111110.

Layer 2 Coding of flags

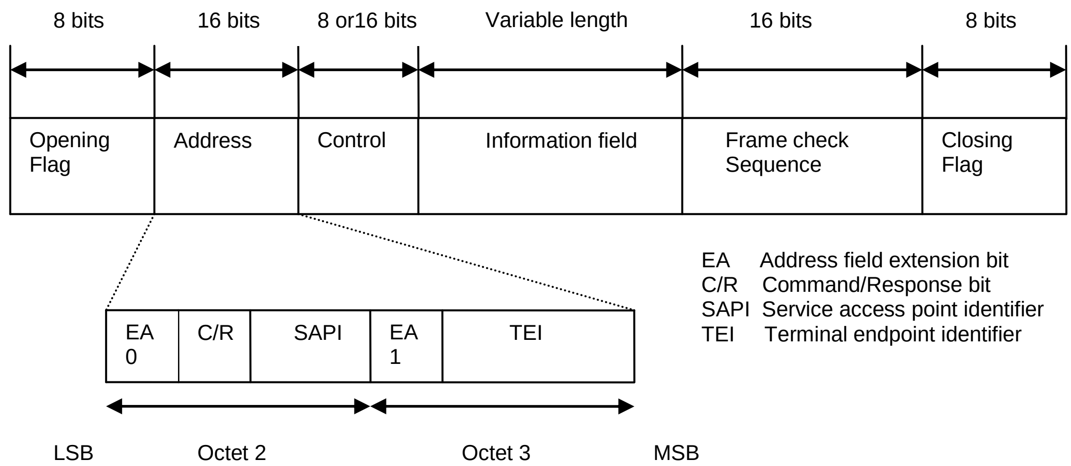


The closing flag of one frame can serve as the opening flag of the next frame. It should be noted that the receivers should be capable of receiving multiple flags.

To avoid the flag pattern occurring elsewhere within the frame, z e r o b i t i n s e r t i o n is used. Whenever five contiguous 1s are detected within the frame (other than within the flags), a zero is inserted between the fifth and sixth 1s (i.e. 01111110 -> 011111010). This zero is removed again on receipt.

The address is two octets long (16 bits) and consists of the following elements as shown below.

Layer 2 Address Field Format



EA is the address field extension bit, which occupies the first bit of each octet of the address field and is used to indicate the address field length. The EA bit is coded as a one in the final octet of the address field and 0 in all the preceding address field octets. So in the case of LAPD where the address field is two octets long, the first EA bit is coded as ‘0’ and the second as ‘1’.

The C/R Command/Response bit identifies a frame as either command or response. The user side shall send commands with C/R bit set to 0 and responses with C/R = 1. The network side does the opposite. The commands are sent with C/R set to 1 and responses with C/R = 0.

As to be able to address multiple devices (TE´s) with the point to multipoint configuration, two addressing values are used. The TEI (Terminal Endpoint Identifier) and SAPI (Service Access Point Identifier). TEI value is dedicated to address the specific physical device. On the other hand, one physical device can be assigned more different TEI´s, but one TEI may not be assigned to more than one terminals.

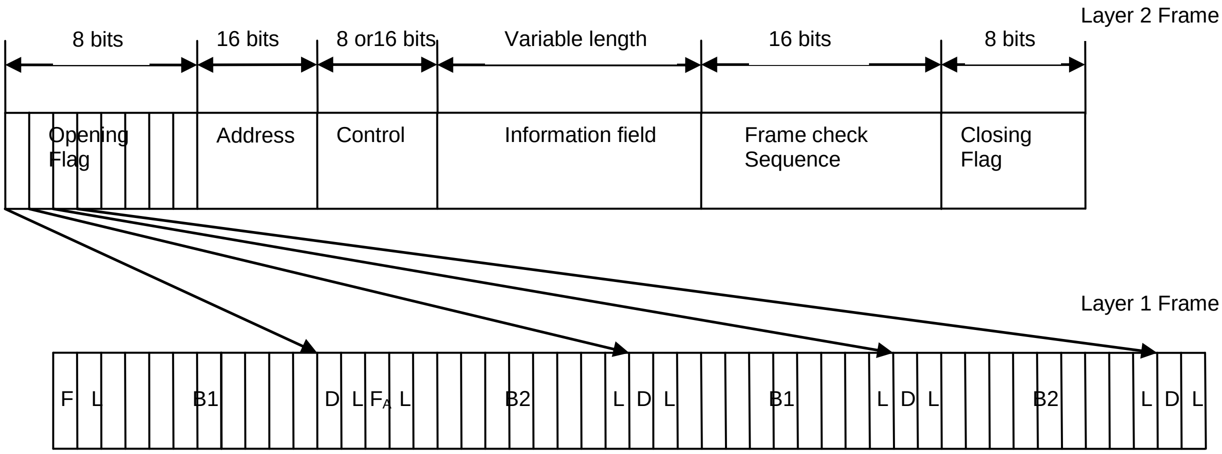
Possible TEI values are 0-63 for “non-automatic TEI assignment user equipment”, 64-126 for “automatic TEI assignment user equipment” and 127 for “broadcast TEI”.

SAPI values identifies requested service which the data in the frame belongs to. The Service Access Point (SAP) is the conceptual interface between two adjacent protocol layers (Layer1 – Layer 2 or Layer2 – Layer 3). Possible SAPI value identifiers are

0	Call control procedures using Q.931 procedures (KCS uses 0)
1	Reserved for packet mode communication using Q.931 procedures
16	Packet mode communication using X.25
32-47	Reserved for national use
63	Layer 2 management functions (i.e. TEI assignment / removal with automatic TEI user equipment).
Others	Reserved for future standardizations

The layer 2 frame is transmitted in the D channel of the layer 1 frame, i.e. each D bit in the layer 1 frame is used to transmit one bit of the layer 2 frame. Hence each layer 1 frame can be used to transmit only 4 bits of a layer 2 frame, and hence a single layer 2 frame is transmitted over many layer 1 frames. See below:

Layer 2 Frame Position



4.3 ISDN Layer 3 Definition

The ISDN Network Layer 3 is concerned with the following functions:

- Establishment of calls across the ISDN Establishment of calls between the calling and called TE´s, including negotiation of B channels, checking service compatibility, supplementary service activation.
- Maintenance of calls across the ISDN Maintenance of the call, including error detection and recovery, call suspension and re-establishment.
- Clearing of calls across the ISDN Clearing of call on completion or due to certain conditions, disconnecting and releasing of B channels.

4.3.1 Layer 3 Differences Between Primary and Basic Rate ISDN

The ISDN primary rate layer 3 protocol is very similar to the ISDN basic rate layer 3 protocol. The differences between the two are summarized below:

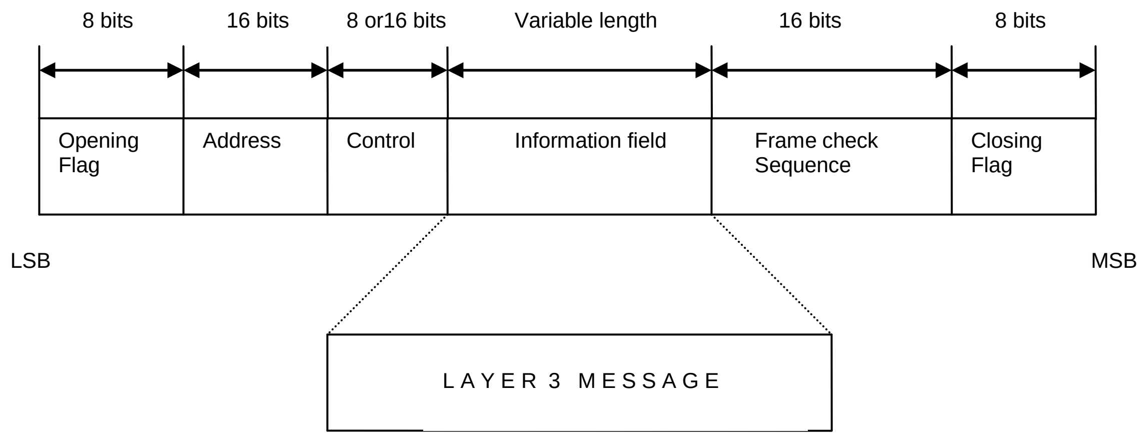
- Primary Rate is point to point, where as basic rate can be point to multipoint
- Primary Rate has 30 B channels or 23 B-channels while basic rate has just 2 B -channels. As a result the Channel ID information element is slightly different.
- Primary Rate does not support Call Rearrangement procedures (SUSPEND and RESUME)

As there is no other difference between basic and primary rate ISDN, the following description is valid for both types of connection.

4.3.2 Layer 3 Message Transmission

ISDN Layer 3 information elements are referred to as m e s s a g e s. Layer 3 messages are transmitted across the TE/NT interface in the information field of layer 2 frames, see below.

Layer 2 Information Frame containing Layer 3 Message



4.3.3 Layer 3 Message Structure

All layer 3 messages are structured into the following information elements, which are discussed in further detail below:

- Protocol Discriminator information element
- Call Reference information element
- Message Type information element
- Additional information elements (as required)

The basic layer 3 message structure is shown in the next drawing below. Each information element consists of an integer number of octets (numbered octet 1, 2 etc.). The eight bits of the octets are numbered bit 1, 2 etc. Octet 1 is transmitted first, starting with bit 1.

LAYER 3 Message Structure

Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	
Protocol Discriminator								Protocol Discriminator IE Octet 1
0	0	0	0	Length of call reference value				Call reference IE Octet 1
Call reference value								Octet 2 .... Octet n
0	Message type							Message type IE Octet 1
Additional Information elements as required								Message type IE Octet 1 .... Octet n

The **Protocol Discriminator** is used to distinguish between user-network call control messages and other types of messages (giving the D channel the capability of being able to support additional communication protocols). The defined protocol discriminator values are given below.

00000000 to 00000111	not used in message protocol discriminator (reserved for user-user information element protocol discriminator)
000010000	ISDN Layer 3 user-network call control messages (Q.931)
00010000 to 00111111	Other layer 3 protocols (including X.25)
01000000 to 01001111	national use
01010000 to 11111110	Other layer 3 protocols (including X.25)

Layer 3 allows the setup of several communications, i.e. calls, over the same layer 2 link using the same TEI, at any one time. The **call reference value** is used to identify a particular call and hence to distinguish between various calls on a particular link.

The call reference value is only applied to the call across the layer 2 link, it is not applied end to end through the network. Hence, for a particular call, different call reference values will be used between the calling TE and its associated NT and the called TE and its associated NT

The call reference value can be a variable number of octets in length (normally between one and three). As a minimum, a primary rate terminal should support single octet call reference values. The length of the call reference, in octets, is defined at the start of the call reference information element (in bits 1 to 4), prior to the call reference value itself.

Call reference values are assigned by the NT or TE responsible for originating call setup across the link (i.e. the TE at the calling TE end or the NT at the called TE end). This same call reference value is then used for the duration of the call (or until the call is suspended). Once the call is complete (or has been suspended), the call reference value is then released, making it free again for a future call.

The TE and NT may potentially both assign the same call reference values to different calls. Hence to distinguish between the two calls in this case, and to distinguish between calls originated by the NT and calls originated by the TE in general, bit 8 of the first octet of the call reference value is used as a flag, the call reference flag. This flag is set to 0 in messages sent from the side that originated the call reference (Originating flag), and is set to 1 in messages sent to the side that originated the call reference (Destination flag). The format of the call reference information element is shown below

LAYER 3 Call Reference Information Element Format

Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	
Protocol Discriminator								
0	0	0	0	Length of call reference value				Octet 1
Flag	Call reference value							Octet 2 .... Octet n
0	Message type							
Additional Information elements as required								

Flag = 0 for messages from originating equipment
= 1for messages to originating equipment

Length = 0 0 0 1 for single octet call reference values
= 0 0 1 0 for two octet call reference values
= 0 0 1 1 for three octet call reference values

The **Message Type** identifies the type of message, i.e. the function of the message being sent (e.g. SETUP, RELEASE etc.). The Message Type is specified as a seven digit code (bit 8 of the octet containing the message type is set to 0). The defined message types are grouped into four subgroups based on their functionality. Bit 7 and Bit 6 are used to indicate the subgroup:

- Call establishment (bit 7 = 0, bit 6 = 0)
- Call information phase (bit 7 = 0, bit 6 = 1)
- Call clearing (bit 7 = 1, bit 6 = 0)
- Miscellaneous (bit 7 = 1, bit 6 = 1)

The remaining bits (bit 5 to bit 1) are used to uniquely identify the message type. The defined message types, and coding used in the message type information element, is defined in the drawing below:

LAYER 3 Message Type Section Format

Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	
Protocol Discriminator								
0	0	0	0	Length of call reference value				
Flag	Call reference value							
0	Message type A	B	x	x	x	x	x	Octet 1
Additional Information elements as required								

A B = 0 0 for Call Establishment
= 0 1 for Call Information Phase
= 1 0 for Call Clearing
= 1 1 for Miscellaneous messages

Call Establishment

```
0 0 0 0 0 0 0 1   ALERTING
      0 0 0 1 0   CALL PROCEEDING
      0 0 1 1 1   CONNECT
      0 1 1 1 1   CONNECT ACKNOWLEDGE
      0 0 0 1 1   PROGRESS
      0 0 1 0 1   SETUP
      0 1 1 0 1   SETUP ACKNOWLEDGE
```

Call information phase

```
0 0 1 0 0 0 0 0   USER INFORMATION
      0 0 0 0 1   SUSPEND REJECT
      0 0 0 1 0   RESUME REJECT
      0 0 1 0 1   SUSPEND
      0 0 1 1 0   RESUME
      0 1 1 0 1   SUSPEND ACKNOWLEDGE
      0 1 1 1 0   RESUME ACKNOWLEDGE
```

Call clearing

```
0 1 0 0 0 1 0 1   DISCONNECT
      0 1 1 0 1   RELEASE
      1 1 0 1 0   RELEASE COMPLETE
```

Miscellaneous

```
0 1 1 0 0 0 0 0   SEGMENT
      1 1 0 0 1   CONGESTION CONTROL
      1 1 0 1 1   INFORMATION
      0 0 0 1 0   FACILITY
      0 1 1 1 0   NOTIFY
      1 1 1 0 1   STATUS
      1 0 1 0 1   STATUS ENQUIRY
```

4.3.4 Layer 3 Call Establishment

Described below are the procedures for establishment of a call between two TEs, the calling Party (the TE initiating the call) and the called Party (the TE receiving the call). Call establishment is often described in terms of the procedures applicable to the calling TE – outgoing call establishment, and the procedures applicable to the called TE incoming call establishment. Both incoming and outgoing call establishment are described below, as well as the relation between the two.

Call establishment can either use E n b l o c or O v e r l a p Sending.

- With **E n b l o c** sending, layer 3 call establishment is not initiated by the TE until the complete called TE address is available (e.g. until complete number has been dialed by the user). In this case the SETUP message will contain the complete called TE address.
- With **O v e r l a p** sending, call establishment can be initiated before the complete called TE address information is available (e.g. before complete number has been dialed by user). In this case the SETUP message will not contain the complete called TE address. The remaining called TE address information will be sent in INFORMATION messages as it becomes available.

Setup procedures are described below for both Enbloc and Overlap sending.

Call establishment is initiated by the Calling Party sending a SETUP message. The SETUP message will contain a call reference value selected by the TE. The SETUP message shall also contain information regarding the required B channel (indicating if a particular channel is required, preferred or if any channel is acceptable).

In the case of En-bloc sending, i.e. the SETUP message contains all the Called Party address information, and if all the Setup information is acceptable, then the NT shall return a CALL PROCEEDING message to the Calling Party, connect the appropriate B channel and forward the call to the Called Party.

In the case of Overlap sending, i.e. the SETUP message does not contain all the Called Party address information, then the NT shall return a SETUP ACKNOWLEDGE message to the Calling and activate the appropriate B channel. The Calling Party will then send the remaining Setup information using INFORMATION messages, until all setup information is complete. Providing the Setup information is acceptable, the NT returns a CALL PROCEEDING message to the Calling Party, and forwards the call to the Called Party.

On receipt of the incoming call at the far end NT, the NT will send a SETUP message to the Called Party. The SETUP message will contain a call reference value selected by the NT, and will contain information on the B channel (indicating a particular channel that may or may not be negotiable, indicating any channel available or indicating no channel available).

If the SETUP message sent to the Called Party does not include all the called number information then the Called Party will transmit a SETUP ACKNOWLEDGE. The NT will then transmit the outstanding information in INFORMATION messages.

On receipt of the SETUP message (and corresponding INFORMATION messages in the case of Overlap Receiving), the Called Party will check that it is compatible with the bearer capabilities indicated in the SETUP message. The Called Party will also consider the B channel information in the SETUP message. If it wishes to select a particular channel it will inform the NT in the next response that it sends to it.

The Called Party may then respond in any of the following optional ways before acceptance of the call:

- Send a CALL PROCEEDING message to indicate call establishment has been initiated.
- Send an ALERTING message to indicate called user alerting has been initiated (user´s terminal has received the call request and is alerting the user), which will be forwarded through the network to the Calling Party
- Send a CALL PROCEEDING message, then send an ALERTING message, which will be forwarded through the network to the Calling Party
- None of the above.

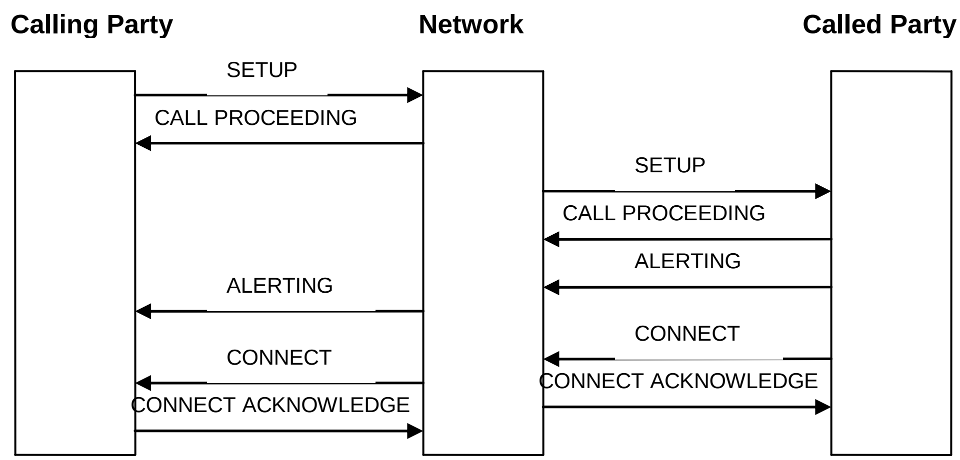
The called Party accepts the call by sending a CONNECT message to the NT. On receipt of the CONNECT message, the NT connects to the indicated B channel, returns a CONNECT ACKNOWLEDGE message to the Called Party and forwards the CONNECT message through the network to the Calling Party.

On receipt of the CONNECT ACKNOWLEDGE message, the Called Party connects to the B channel and is taken to state Active. On receipt of the CONNECT message by the Calling Party, the Calling Party will enter state Active and may optionally return a CONNECT ACKNOWLEDGE message to the NT.

4.3.4.1 Connecting the Call

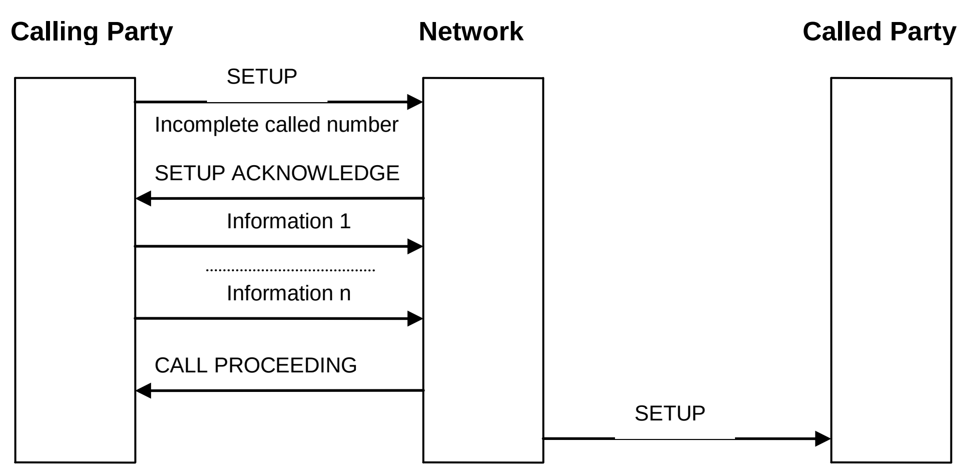
In the following example below, the calling party starts the sequence by sending a SETUP message to the network. In the SETUP message, the user sends the network the information that the network needs to connect the call. Some examples of such information are the desired Bearer capability, the identity of the called party and the B-channel that the user's terminal suggests to be used for this call.

En-bloc Setup Procedure



In the next drawing the SETUP procedure of "overlap" sending is described compared to the above method of "en-bloc" sending:. Only the SETUP will be explained, the remaining part is identical to the En-bloc sending procedure.

Overlap Setup Procedure

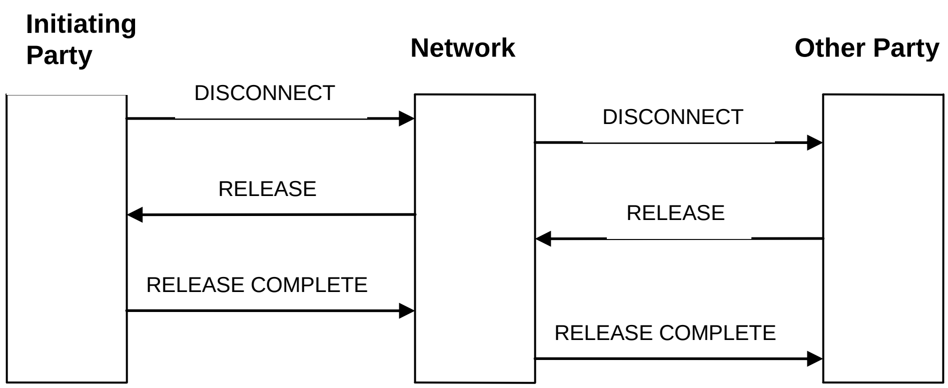


4.3.4.2 Disconnecting the Call

The procedure starts when one party's terminal sends a DISCONNECT message to the network and disconnects itself from the B-channel. The network returns a RELEASE message to the party that initiated the disconnect and initiates procedures to have a DISCONNECT message sent to the other party.

The initiating party's terminal completes the exchange of messages with a RELEASE COMPLETE message. At this time, the network releases the B-channel.

Disconnecting Procedure



5. Bearer Capabilities and Service Indicators

Bearer capabilities is a mechanism by which a user can request a corresponding bearer service for the application protocol that he is using in a B channel.

Strictly speaking, the Bearer Capability is one of three mechanisms that are used to communicate this information. However, the other two mechanisms (called Higher Layer Compatibility - HLC and Lower Layer Compatibility - LLC) are always used in conjunction with Bearer Capabilities, so they are often known collectively as bearer capabilities. The HLC and LLC are used to provide additional details about the B channel protocol.

Although the ISDN doesn't always need to know what protocol is in use in the B channel, there are circumstances where this information is useful.

For instance, if you make a telephone call over ISDN, it is useful to tell the network that this is a voice call so that it can connect your call with an analogue telephone in the PSTN. If the ISDN network thinks that your B channel contains a protocol unknown to it, then it can only connect your call directly to another ISDN line. On the other hand, the network is allowed to transcode the voice application data into a more efficient data format (even with information loss), or to discard the “silence” between words to further increase efficiency. This is fine for voice but not so fine for some other voice-band data transmissions like fax.

The receiver of the call can also see the Bearer Capabilities requested by the call originator when a call is offered to him by the network.

5.1 Usage of Bearer Capabilities (EuroISDN and QSIG)

Understanding Bearer Capabilities and their implications becomes significant when diagnosing problems, particularly when using diagnostic traces.

An example of the problems that can occur involves fax calls not being received correctly because there are two possible types of Bearer Capability for a fax call.

- **3.1kHz analogue call -** This often happens if the call has been originated inside the PSTN. The ISDN has no way of knowing exactly what type of equipment (telephone, fax, modem, etc.) placed the call, so it uses this "catch-all" bearer capability.
- **Fax Group 3 call** - This is often used if the originator sends with a fax machine that is directly connected to an ISDN network.

For the call to be successful, the chosen Bearer Capabilities must be supported by all equipment throughout the path taken by this call across the network. If any single piece of equipment does not support the requested Bearer Capabilities, the call will not be connected

5.2 KCS Bearer Services (EuroISDN and QSIG)

3,1kHz audio for outgoing calls coded in the outgoing SETUP message

3,1kHz audio and ***Speech*** for incoming calls. If the incoming SETUP message contains other bearer capability than this one, the call will be rejected.

Note: Speech bearer service is forbidden to use with facsimile terminals. Nevertheless, it has been added due to problems with various PABX systems.

5.3 KCS Tele Services (EuroISDN and QSIG)

Remember the following from the previous chapters: What really separates tele services from bearer services are the higher layer compatibility, end-to-end functions.

For outgoing calls from KCS with the standard configuration, no higher layer compatibility element (HLC) is coded within the outgoing SETUP message.

Facsimile terminals should use HLC information element coded “facsimile G3” for outgoing calls. A list of possible used HLC’s can be configured with ISDN config line 251. It has been made configurable because there are wrong configured PABX systems or ISDN a/b terminal adapters (TA) that will reject these calls with “facsimile G3” HLC information element.

Facsimile G3 teleservices for incoming calls: if HLC information element is available within the incoming SETUP message. If HLC is available but coded differently, ignore the call. If not available, not checked at all – accept the call only according to the bearer service compatibility.

	Bearer services	Tele services
Incoming call	3,1kHz audio Speech	Facsimile G3
Outgoing call	3,1kHz audio	None (default) FAXsimile G3 (config)

5.4 Usage of Service Indicators (1TR6)

ISDN services with 1TR6 are coded differently as with EURO-ISDN. The only (mandatory) information element (called W-Element) Service indicator is used, services are not explicitly divided into bearer- and Tele services. Service indicator information element consists of two parts: ***Service group*** and ***Service***.

5.4.1 Service Indicator Coding

Service group “***a/b Service / Fax Group 3”*** for outgoing calls (by default).

Note: Service indicator for outgoing calls can be optionally configured with config line 251 due to problems at various installations with this default setting.

Service group “***a/b Service / Fax Group 3”*** or Service group ***speech***, Service ***speech analogue*** or ***speech***, ***3.1 kHz audio*** for incoming calls. If the incoming SETUP contains the Service indicator with different coding, the call will be ignored.

	Service indicator
Incoming call	a/b service / facsimile G3 a/b service / Speech analogue a/b service / Speech 3,1kHz
Outgoing calls	a/b service / facsimile G3 (default)

5.5 Higher/Lower Layer Compatibility Settings (EuroISDN and QSIG)

The Setup of the HLC Info element used from KCS and defined within ISDN config position 251 (for outgoing FAX) and ISDN config position 297 (for outgoing VOICE) is defined as follows

:00 FF FF FF FF FF ,251 (or 297)

Position 1 …6, EuroISDN and QSIG service code identifications, Bit values, explanation see below

0 0 0 0 0 0 0 0

Bits 0...5 are used as HLC code
- 00 …. No HLC
- 01 …. Telephone
- 04 …. Facsimile

Bit 6 is used as bearer capability / Lower Layer Compatibility
- 0 …. 3,1kHz analogue
- 1 …. speech

Bit7 is used to define the retry in case info code 300 has been received
- 0 …. Do not initiate a retry
- 1 …. Initiate a retry with next position within config line

ISDN config line 251/297 holds a maximum of 6 values which can be used for the KCS setup message. The default value has been defined as 00 FF FF FF FF FF for FAX and 01 FF FF FF FF FF for VOICE. This means that KCS initiates a SETUP message with “no HLC” for FAX and a SETUP message with “HLC telephone” for VOICE and if this is not successful, sending is aborted immediately.

Service code 00 is used to suppress the HLC info element. According to various recommendations, such calls should be answered from every apparatus at the distant side that may be compatible, e.g. telephone, fax and modems. Nevertheless, in some countries, calls without a HLC info element are not accepted by the local exchange.

Supported combinations of HLC info elements with KCS are:

00 01 04 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, do not use LLC frames, do not make any retries if receiver side aborts the call with ISDN info code 300

40 41 44 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, use additionally “speech” bearer capability within LLC frames, do not make retries if receiver side aborts the call with ISDN info code 300

80 81 84 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, do not use “speech” bearer capability within LLC frames, make retries if receiver side aborts the call with ISDN info code 300

C0 C1 C4 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, use additionally “speech” bearer capability within LLC frames, make retries if receiver side aborts the call with ISDN info code 300

The definition of LLC (Lower Layer Compatibility) and LineServer load balancing used from KCS and defined within ISDN config line 286, 5th position, is defined as follows

:01 00 00 00 00 00 00 00 ,286

Position 5, sending complete indicator for outgoing setup and LLC compatibility definition and Line Server 1 incoming FAX/VOICE load balancing option

Bitvalue: 0 0 0 0 0 0 0 0

Bits 0 and 1 are used as sending complete / LLC
- 00 …. use sending complete indicator, no LLC compatibility
- 01 …. no sending complete indicator, no LLC compatibility
- 02 …. use sending complete indicator, use LLC compatibility
- 03 …. no sending complete indicator, use LLC compatibility

Bit 7, the MSB, is used for LineServer Load Balancing
- Bit 7=0 …. Load Balancing enabled (00, 01, 02, 03)
- Bit 7=1 …. Load Balancing disabled (80, 81, 82, 83)

The sending complete indicator can be disabled by setting the 5th position of config line 286 to 01 or 03. By disabling the sending complete indicator, the ISDN channel operates in the so called “Overlap sending mode” – a special Layer 3 functionality.

This state exists for an outgoing call when the user has received acknowledgement of the call establishment request which permits the user to send additional call information to the network in overlap mode.

It is only used with an Alcatel 4400 PABX, which causes problems while operating with the “sending complete indicator”. So keep in mind that whenever an Alcatel 4400 PABX has to be connected to KCS, this config line has to be set accordingly.

6. ISDN Supplementary Services

TCOSS currently supports the following ISDN supplementary services:

1) Multiple subscriber number (MSN)

2) Direct dialing in (DDI)

3) Advice of charge (AOC) (for EuroISDN only)

4) Call Diversion (CFU, CFB, CFNR, CD) (for EuroISDN and QSIG only)

5) Call Transfer (CT) (for QSIG protocol only)

6) Path Replacement, in conjunction with CT (PR) (for QSIG protocol only)

7) Message Waiting (MWI) (for QSIG protocol only)

Supplementary services are typically not available by default, they must be ordered from the PTT or local PBX administrator.

6.1 MSN – Multiple Subscriber Number

Generally supported only with point to multipoint ISDN access (station line access in the USA), where more different subscriber numbers can be configured for each UIF module. Normally, up to ten numbers with last digit variable are used, i.e. 5158610, 5158611, ..., 5158619. With 1TR6 protocol, this service is called „Endgeraete Auswahl Ziffer (EAZ)“ or „end terminal selection number“.

With EuroISDN it is possible to use completely different numbers for one access, i.e. 5158610, 66133 etc.

If MSN is configured for specific ISDN point-to-multipoint access, the whole number or at least the variable part of own number, which was dialed by the originating user, is given to the destination user in the incoming SETUP message.

If configured in the line 252 (‘1’ = MSN mode), config lines 254-283 must contain at least one of the several valid numbers for the access. During the incoming call, the own number provided by the PTT in the incoming SETUP message is compared with those configured in line 254-283. Only if a match occurs, the incoming call is accepted - otherwise, it is ignored. However, this compare is made only if the own number is present in the incoming SETUP message. If not, such an incoming call would be accepted.

Example:

The own ISDN number is 5158610 .. 5158619 (last digit is variable). If the local exchange sends the complete number the configuration 1 must be used. Otherwise if only the extension is send configuration 2 must be used.

config line 235 FXI$ FXI$ FAX$, config line 236 :01 , config line 252 “1”

```
                configuration 1          configuration 2
config line 254    ‘15158611=1   ,          ‘11=1   ,
config line 255    ‘15158612=2   ,          ‘12=2   ,
config line 256    ‘15158615=3   ,          ‘15=3   ,
config line 257    ‘15158616=T10 ,          ‘16=T10 ,
```

KCS answers incoming calls to extensions ‘1’, ‘2’, ‘5’ and ‘6’ only. They are distributed to number “FXI$1”, “FXI$2” and “FXI$3”. Upon calling extension „6“ a DTMF prompt will be activated (value after „T“ is time-out in seconds). This allows input of any inbound code or server command without restrictions. The decision if the whole or only the variable part of the number is provided is made by the local Exchange (LE).

6.2 DDI – Direct Dialing In

The functionality of the DDI (Direct Dialing In) supplementary service is the same as with DID (Direct Inward Dialing) used with the conventional UTF module. In the fact it’s nothing more than the delivery of the dialed called party number – or at least a part of it (extension) - to the destination terminal.

It is generally supported only with point-to-point ISDN access, dedicated mainly for PBX devices to be able to switch the incoming call to one of the connected terminals (telephones). For this reason the whole own number or at least the extension part of the number is sent to the destination user in the incoming SETUP message.

Note: The PTT or local PBX technician has to provide the information how incoming calls are handled. This can be as the complete number information or only its extension part

But there are substantial differences how the DDI functionality works in different countries:

1) Austria and Germany

   These are the only two countries with very open numbering plan: telephone numbers typically consist of a public number part and of the variable part (in German “Nachwahl”). It is even possible that one company may have telephone and fax numbers of different length (like +43-1-86353-123 and +43-1-86353-8123).
   Once the caller has dialed the public part of the number, the PSTN recognizes the public part of the number is complete and establishes the connection to the destination. The variable part of the number comes digit by digit afterwards as the caller dials them on the phone set.

2) Rest of the world

   In all other countries all telephone numbers on the same location must have the same length:
   thus all of company extensions – voice and fax – are true public numbers and it is not possible to dial any digit more. Therefore the caller must at first dial the whole number, the PSTN recognizes the number is complete and establishes the connection to the destination.

   The dialed number (either complete or only the extension part) comes *enbloc* along with an incoming SETUP message to the destination terminal.

Configuration of DDI mode is done via config line 252 by setting the value to ‘2’ (except for the USA and Japan, see below).

Example:

The “basic” own ISDN number is 6621045, 3 digits DDI number is used (i.e. 123). If the local exchange sends the complete number the configuration 1 must be used. Otherwise if only the DDI information is send configuration 2 must be used.

config line 235 FXI$ FXI$ FAX$, config line 236 :03 , config line 252 “2”

```
                configuration 1   configuration 2 (not mandatory)
config line 254    ‘16621045˜=˜ ,     ‘1˜=˜ ,
```

KCS answers all incoming calls. Incoming calls to numbers 6621045123, 6621045124 and 6621045125 are distributed to numbers FXI$123”, “FXI$124” and “FXI$125”, respectively. The decision whether the whole or only the DDI part of the number is provided is made by the local Exchange (LE).

Please note: Configuration 2 in the example above shows a 1:1 DDI translation and is therefore not necessary to setup. Simply leave the content of config line 254 empty. It has been shown here due to compatibility reasons.

6.2.1 DDI in the USA and Japan

Due to an ISDN layer 3 protocol differences it is necessary to activate the “DDI” by setting the config line 251 to ‘1’ as normally used for MSN.

Background:
The US and Japanese ISDN protocols do support only the enbloc receiving procedure (the complete dialed number comes in with the 1st incoming SETUP message). And it is functionally the same as with MSN configuration.
On the other hand, configuring “DDI” may invoke the overlap receiving procedure as dialed numbers are generally not marked as complete in the incoming SETUP message: and this may lead to rejecting of the call.

6.3 AOC – Advice of Charge

For the advice of charge counting both the D and E variants (AOC-D and AOC-E) with charging units and currency units are supported by TCOSS. The UIF module has the possibility to use the fee information (advice of charge =AOC) from the local exchange for cost accounting.

What is the difference between AOC-D and AOC-E messages:

AOC-D	ISDN Advice of Charge information is sent during a call. The message is sent periodically by the network to subscribers of AOC during-call services.
AOC-E	ISDN Advice of Charge information is sent at the end of a call. The message is sent periodically by the network to subscribers of AOC end-of-call services.

Requirements:

- The AOC must be provided by the local exchange.
- Cost accounting with AOC for UIF cannot be used together with cost accounting for UTF and UTT channels. In that case costs for the UIF channels must be calculated in the same way as with an UTF channel.

If both UTF and UIF modules should be used with cost accounting, the user module type for the UIF must be changed. This will be done by setting config line 3 of the ISDN channel to ‘T,. By setting this config line 3 to “T”, we define that the ISDN channel is treated by TCOSS as a TELETEX channel. With that tricky setting, the kk99 cost accounting file can be used for the ISDN AOC calculation.

In that specific case, cost accounting must not be used with UTT channels (UTT not supported anymore) and all lines in the kk99 used for TELETEX must be removed!.

Cost accounting must be activated with config line 8 of the ISDN channel which should calculate costs. The actual type of received charging unit information depends on the local PABX or the exchange. The following list gives an overview.

Exchange or PABX transmits	**Behavior of the UIF module**
No AOC information	No costs are reported
Charging units	Charging units are reported
Currency units	Currency units scaled by configuration are reported
Currency and charging units	Undefined

- Currency units are not supported with 1TR6 protocol
- In Austria only currency units are available
- In Germany both currency and charging units are supported with EuroISDN lines. Since only one type can be used on a line the required type can be configured at the local exchange.
- If both currency and charging units are available, currency units should be preferred.
- KCS expects that either currency or charging units are received on the same line. Therefore it is not required to configure the AOC type. But currency values are scaled as defined within config line 253.

Config line 253, 1 position, Currency units scaling

‘0’ report value in 1/1000 units of currency
‘1’ report value in 1/100 units of currency
‘2’ report value in 1/10 units of currency
‘3’ report value in units of currency
‘4’ report value in 10 units of currency
‘5’ report value in 100 units of currency
‘6’ report value in 1000 units of currency

For each send attempt the count of charging units or scaled currency unites reported by the local PTT during the call clearing phase is returned as the count of 64-bytes frames transmitted. That is why the calculation of fees for the UIF module in kk99 should be based only on the parameter P4.

Formula: P4 = (costs for 1 AOC unit) x 1600.

P4 definition: fee per 1024 characters (in 1/100 units). The cost calculation is based on the number of 64 byte frames transmitted. 1024 characters correspond to 16x64byte frames. The information is read from parameter CHCNT which is used to inform the TAM about the number of transmitted 64 byte units

Example for charging units:

1 charging unit in Germany costs 0,23 DM. Costs should be shown in the kkxx file as multiples of 1/100 DM (=1pfennig). Therefore set config line 3 to ‘T’, config line 253 will be ignored for charging units.

kk99 File:

```
1,0,0,1,0,36800   p4 = 23 * 1600 = 36800
0=1
1=1
2=1
3=1
4=1
5=1
6=1
7=1
8=1
9=1
```

Example for currency units:

Costs should be shown in kkxx files as multiples of 1/100 DM (=1pfennig). Therefore set config line 3 to ‘T’, set config line 253 to ‘1’.

kk99 File:

```
1,0,0,1,0,1600    p4 = cost for 1 AOC unit x 1600
0=1
1=1
2=1
3=1
4=1
5=1
6=1
7=1
8=1
9=1
```

Please keep in mind that some PABX devices, e.g. Siemens Hicom 300, do not to provide the final charging information with AOC-D service when the call is terminated. Therefore KCS evaluates also charging information sent during the call now and does not require the call-termination charging information anymore. This new feature works automatically and does not require any configuration changes.

Note:

When Currency Units variant of AOC is used, the right currency scaling factor must be configured by the config line 253. It is recommended to set it to the same scaling factor that is used by the network to submit the charging information. Typical scaling factors used in the field are:

Germany: 1/100 DM (1 pfennig), therefore set conf. line 253: ‘1’ (report value in 1/100 currency units)

Austria: 1/1000 ATS, therefore set conf. line 253: ‘0’ (report value in 1/1000 currency units)

If you are not aware of the scaling factor used by the network, do the following:

Set config line 253 to ‘0’ (report value in 1/1000 currency units) and send one short local outgoing message that would cause 1 charging unit and check the costs by TCfW (e.g. in Germany, one charging unit costs 12 pfennig, you would see 120 within your outbox. You may increase the scaling factor to 1/100 in order to get the cost as 12 pfennig).

If you would set the scaling factor “too high” (e.g. conf. line 253 to ‘3’ in Germany) some amount of charging information could be lost since KCS uses simple integer division on converting the charging information (e.g. one local call in Germany that costs 12 pfennig would be converted to 12/100 = 0 charging units). If you try a local outgoing call and after the call has been terminated you see costs set to 0 within your TCfW outbox, there are two possibilities:

1. The network (PTT or PABX) does not provide charging information at all
2. You have set a “too high” scaling factor with conf. line 253. Try the value ‘0’ (1/1000 currency units).

6.4 Call Diversion Supplementary Services

6.4.1 Basic Terms

The Call Diversion supplementary services comprise the following services:

1) Call Forwarding Busy (CFB)

2) Call Forwarding Unconditional (CFU)

3) Call Forwarding No Reply (CFNR)

4) Call Deflection (CD)

The Call Forwarding service enables a served user to have his incoming calls redirected (automatically by the service provider) to another user upon one of the following conditions:

1) Served user is busy (CFB)

2) Server user is not responding (CFNR)

3) Unconditional – calls are redirected independently of served user’s status (CFU)

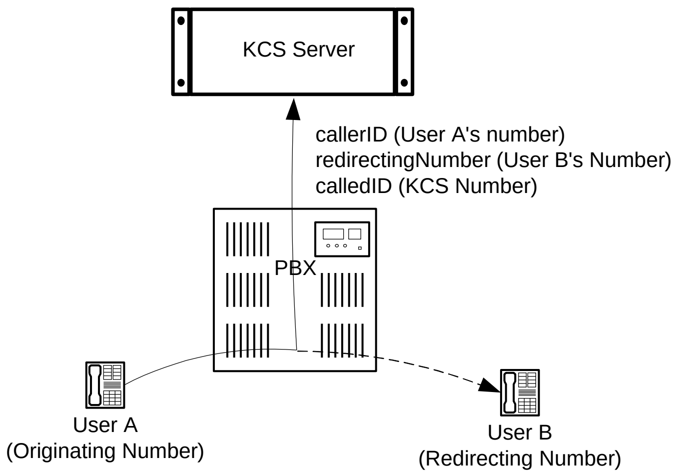
The destination number (the diverted-to-user’s number) is being entered during Call Forward activation phase either by the server user himself, or centrally for all served user’s by the PTT or local PBX administrator.

The Call Deflection service permits a served user to respond to his incoming calls by (manually) requesting redirection of that call to another user. This request is only allowed before the served user has answered the call.

The served user's ability to originate calls is unaffected by the Call Diversion supplementary service.

6.4.2 Redirecting Number

Assume user A calls user B, but the PBX serving the user B recognizes one of the Call Forwarding conditions CFU, CFB or CFNR and redirects the call to the KCS Server:



If Call Diversion services were activated for the KCS line, the PBX delivers also the "Redirecting Number" – the originally dialed number along with a redirected call setup.

The "Redirecting Number" may be very effectively used for both fax and voice PBX integrations (see below).

KCS Server supports the **Redirecting Number** according to the EuroISDN call Diversion (ETS 300 207) and QSIG Call Diversion supplementary services.

Note: The most of PBX systems does not support the Redirecting number functionality via EuroISDN trunks. Therefore it is recommended to use QSIG protocol if this function is requested.

6.4.3 Original and Last Redirecting Number of a Multiple Call Diversion

Assume user A calls user B, but the PBX serving the user B recognizes one of the Call Forwarding conditions CFU, CFB or CFNR and redirects the call to (3rd) user C.
User C is also not responding and the call is further redirected to the KCS Server.

In this case LS1 server receives two "Redirecting numbers" along with the incoming call setup:

1) The *Original Redirecting number* as the number that was really originally dialed by the caller

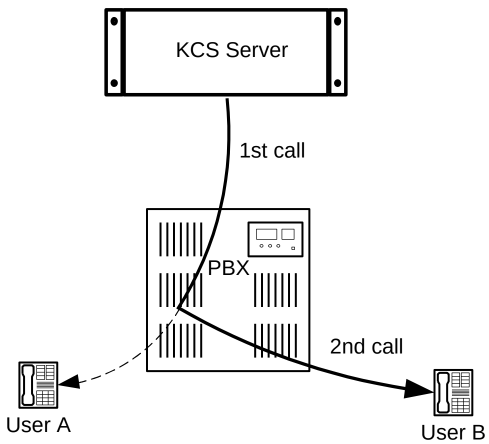
2) The *Last Redirecting number* as the last number that redirected the call towards KCS Server

LS1 server can use either the *Originally Called Number* or the *Last Redirecting Number* for Voice call inbound routing (configurable).

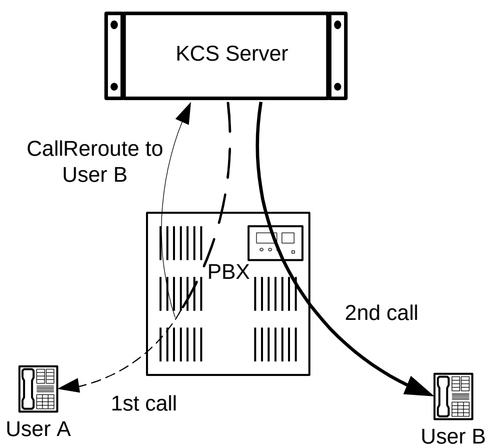
6.4.4 Network Routing Algorithm

Assume KCS server calls user A but this call is being redirected to user B due to one of the Call Forwarding conditions CFU, CFB or CFNR.

With *Forward Switching* algorithm the PBX establishes a 2nd call to user B, informs the KCS server about the performed call forward and simply joins both connections together:



With *Rerouting* algorithm the PBX only informs the originator – the KCS Server – by the means of Call Rerouting command that this call has to be rerouted to user B:



Upon receiving the Call Rerouting command the KCS Server disconnects the 1st call and establishes the new (2nd) call to user B.

6.4.5 Call Diversion Supplementary Service for FAX Calls

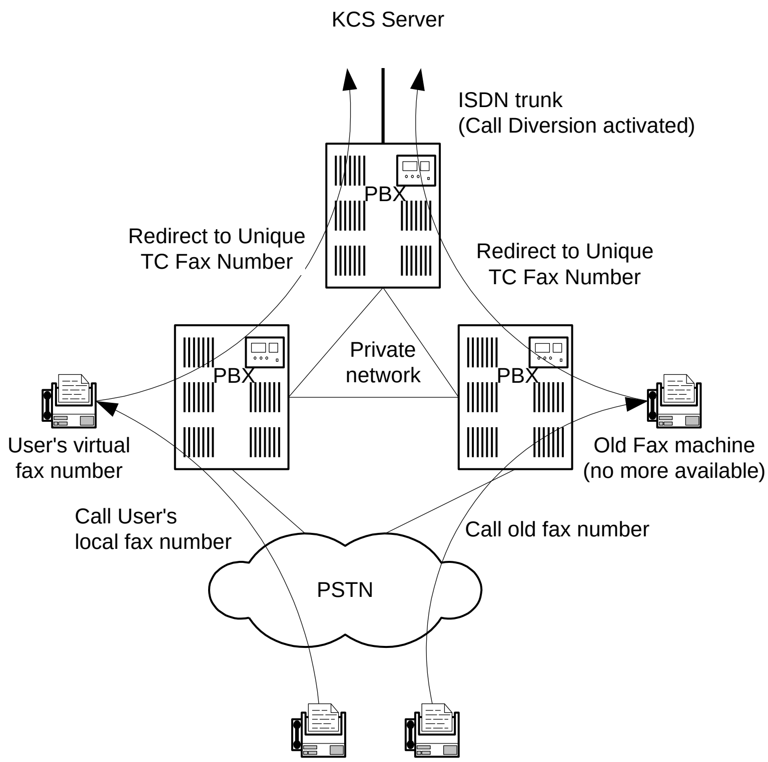
The Call Diversion supplementary service may be interesting for customers having an own PABX network that interconnects several offices.

<u>Note that using of Call diversion services for fax calls cannot be combined with Call diversion service foe voice calls!</u>

Two scenarios are possible:

1) The customer assigns virtual fax extensions for users in local branch offices. In the local PBX these virtual numbers are CFU-forwarded to a unique KCS fax access number.
   From public network, these virtual fax numbers can be dialed using local area code of the branch office.

2) The customer wants to replace old fax machines in the branch offices but wants to keep old fax numbers Fax machines are removed but their extensions are CFU-forwarded to the unique KCS fax access number.



The only configuration task on the KCS side is to recognize redirected calls and use the Redirecting Number use it for inbound routing instead of the called number (DDI) that may be even the same for all incoming calls.

The definition of redirected number functionality used from KCS and defined within ISDN config line 286, 8th position, is defined as following:

:01 00 00 00 00 00 00 00 ,286

Position 8, redirecting number functionality enabled or disabled

00 …. Redirecting number disabled
01 …. Redirecting number enabled, if available with an incoming call it is used for inbound routing instead of DDI information, if not available, then DDI is used for inbound distribution

6.4.6 Call Diversion Supplementary Service for VOICE Calls

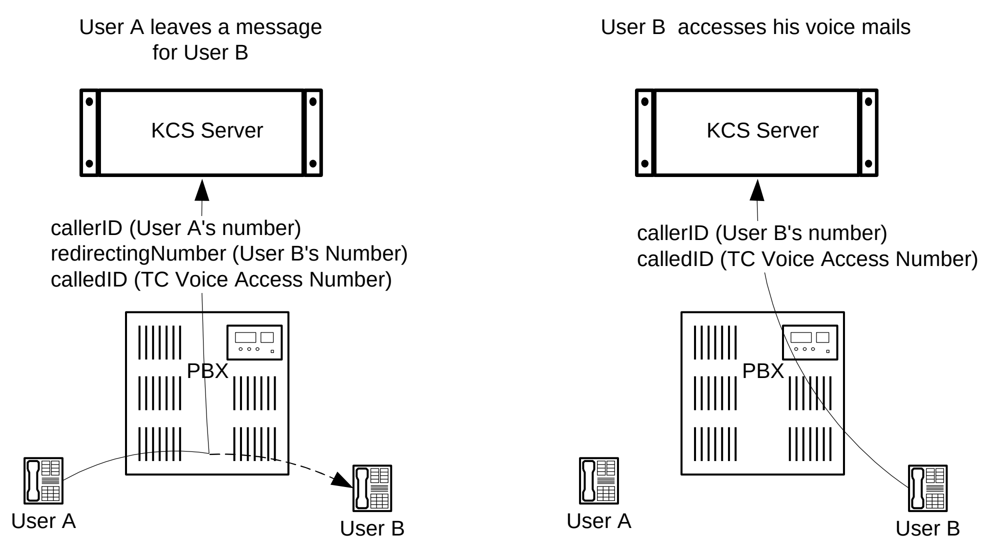
Call Diversion supplementary service can be used to implement the Voice KCS-PBX integration using the single Voice Access Number (allocated for the KCS trunk on the PBX side).

<u>Note that using of Call diversion services for voice calls cannot be combined with Call diversion service for fax calls!</u>

The only assumption is that all telephones company-wide must be forwarded to this same unique Voice Access Number.

1) If a voice call is redirected to KCS Voice Access Number, KCS recognizes that this is a redirected call and uses the Redirecting Number for inbound routing instead of the called number (the caller will be routed into the voice box determined by the Redirecting Number)

2) All users access their mailboxes as usually via the same Voice Access Number



The advantages of using Call Diversion for the Voice Integration:

1) No separate number range necessary for VOICE mailboxes on the KCS server

2) Call forwarding activated to the same Voice Access Number – easy administration

If Call Diversion services would be not supported by the PBX, the only alternative would be to use a unique Voice box extension. For example, for users with telephone extensions 1000, 1001 … call forward may be setup to extensions 2000, 2001 … assuming the 2xxx number range will be routed to the KCS server. The KCS server will receive this "forward to" extension as the **called party number**, (calledID) and performs inbound routing according to this number (calledID is the inactive voice number in the user's profile).

This method is technically simple but not suited very well for the real world: a complete unique number range would be occupied in the PABX only for internal voice mailboxes.

The prerequisites for the "Redirecting Number" functionality are:

1) The PBX must support Call Diversion supplementary services. Although KCS supports this service also for EuroISDN, the preferred way of integration is the QSIG protocol as the most of known PBX systems do support Call Diversion services only for QSIG trunks.

   For a list of PBX systems please refer to the Model 305 PBX Integration Manual.

2) Usage of redirecting number for voice calls must be configured within the UIF module by changing the third position of config line 295.

:00 00 00 00 00 00 00 00 ,295

Position 3, redirecting number functionality for VOICE calls enabled or disabled

00 …. Redirecting number for VOICE calls disabled
01 …. The last Redirecting number for VOICE calls enabled
02 …. Reserved
04 …. The original Redirecting number ("The originalCalledNr")

6.4.7 Testing the Redirecting Number Functionality for Voice Calls

Within this chapter you will learn how the redirecting number functionality works with a LS1 telephone engine and how to test this functionality and additionally how to read the trace file for redirected incoming VOICE calls.

Setup both TCOSS and Voice servers as usually but don't forget to enable the redirecting number functionality (ISDN config line 295, position 3 must be set to 01 and QSIG protocol must be setup on both the KCS and PABX side). Additionally create a TR-trace tup channel within the wconfig program which has to be activated on the DSP0.

During the Voice server setup, define a unique extension number as the „Voice Access number" (this is the number dedicated to access any voice mailbox, for example „99".

Setup this number to be a Voice number in the ISDN module's number conversion table (config line 254 - 283).

```
Config line 254: 138=V99  ** has a special meaning
```

Please note, in our example above you must dial the telephone number 38 to access the KCS Voice Server and to listen to the standard VOICE mail procedure *"This is TOPCALL Voicemail – Please enter the number of your mailbox….".* The telephone number *38* is rerouted to a VOICE number 99. Therefore the access number for the VOICE Server must be defined as 99.

```
HKEY_LOCAL_MACHINE\SOFTWARE\Topcall\TCECP\TCVMAIL\AccessNumbers REG_SZ 99
```

The next step is to call the VOICE Access Number directly from any other internal telephone extension or from an external telephone to verify whether the Voice server responds with the proper prompts – *"This is TOPCALL Voice mail, please enter the number of your mailbox ...".*

Now activate the call forward destination – possibly for all 3 forward types – cfu, cfb and cfnr – for one telephone extension (for example "49") to the Voice Access telephone number ("38")

Setup a user profile for the extension „49" via TCFW and define an inactive VOICE address with the service VOICE

Active	No address no.	Service	Number:
X	1	TOPCALL	TCTECH,
	2	VOICE	49,

Setup is now ready so the next step is to call the telephone extension "49" from any other telephone, we used telephone extension "40" and let the call be forwarded to KCS (for example, don't answer the call - cfnr)

If you hear now the voice prompt (*"The person with extension 49 cannot answer you call"*) asking you to leave a message – redirecting number **WORKS**.

If you hear the previous VOICE prompt (*"This is TOPCALL Voice mail, please enter the number of your mailbox ...".*) then the redirecting number **DOES NOT WORK**. In this case the redirecting number has not been delivered to KCS from the PABX

What to do if the redirecting number does not work:

- Double check with the PABX technician that the QSIG protocol is configured on the PABX and on KCS on all involved ISDN lines
- Configure a Trace module within the WCONFIG and trace several call forward attempts (as described above) and check them by yourself, please read on how to do this.

After the Trace tup channel has been activated simply call the telephone extension where the cfu, cfb and cfnr have been activated for – for example “49“. Call once from another telephone extension and afterwards open the TCOSS trace file

Calling telephone number 40
Called telephone number 49 (This telephone has a condition cfu set-up to telephone extension 38 which is the VOICE server. This number 38 is re-routed to V99 within the ISDN number conversion table

```
[TCOSS] N3/T2A .030=<< L=38 A=02FE,C=03 Setup(01):
          A1= 4=9090A3 18=89 1E=8583 6C=01803430 70=813338 74=0100803439<

Remember:  Info element 6C = calling party
           Info element 70 = called party number
           Info element 74 = new info element and shows redirecting number

[TCOSS] N3/T20 .001=       << SETUP(24)     CR = 1 L3ST=1
[TCOSS] N3/T1F .002=N_CONN_IND(24)  BCHAN=1
[TCOSS] N3/T1F .000=          CALL_PROC(24) >>
[TCOSS] N3/T1D .000=>> L=3 A=00C3,C=7F<
[TCOSS] N3/T2C .074=CI Int. Slot=2 SD S=2(Active) C=0
[TCOSS] N3/T2C .002=CI Int. Slot=2 AI8  Layer 1 is ready     S=2(Active) C=0
[TCOSS] N3/T23 .797=IFX DDI Number=38 OrigNum=40 RedirectingNum=49
[TCOSS] N3/T1D .128=>> L=3 A=00C3,C=7F<
[TCOSS] N3/T2A .016=<< L=3 A=00C2,C=73<
[TCOSS] N3/T1D .083=>> L=11 A=00C3,C=0000 Call-Proc(81):  18=89<
[TCOSS] N3/T1F .000=          CONNECT(24) >> CHIP=0
[TCOSS] N3/T2A .027=<< L=4 A=00C2,C=0102<
[TCOSS] N3/T1D .000=>> L=8 A=00C3,C=0200 Connect(81): <
[TCOSS] N3/T2A .020=<< L=4 A=00C2,C=0104<
[TCOSS] N3/T2A .050=<< L=11 A=02C2,C=0004 Conn-Ack(01):  18=89<
[TCOSS] N3/T1D .000=>> L=4 A=02C3,C=0102<
[TCOSS] N3/T20 .000=       << CONN ACK(24)  CR = 1
[TCOSS] N3/T1F .003=N_CONN_CONF(24) BCHAN=1
[TCOSS] N3/T23 2.00=GETNUM (type 1) 38 => V99 ** ISDN number conversion
[TCOSS] N3/T23 .000=CallerId: Converted 40 => 40
[TCOSS] N3/T23 .000=CallerId: Converted 49 => 49
[TCOSS] N3/T23 .121=Voice Call with CalledId=99, CallerId=40 RedirectingNum=49

[TCOSS] N3/T23 .001=tcTel[0,0,0] send REQ_INIT[0] STR_LineId="", STR_SrcAddr="40",
STR_DestAddr="49", BOOL_Realtime=1, L_Formats=(SET_Format=(INT_FormatType=1,
RAW_Format=<20 Bytes>)), 0_1=<0 Bytes>
[TCOSS] N3/T23 1.10=tcTel[0,C001001A,0] receive Response[11] Ok
[TCOSS] N3/T23 1.10=tcTel[0,C001001A,0] receive REQ_START[13] INT_StreamMode=2,
INT_PositionMsec=0, INT_Sequence=1, SET_Format=(INT_FormatType=1, RAW_Format=<20
Bytes>)
```

If you find trace lines as attached below, which show that the Redirecting number field is empty then the PABX simply does not forward this information to KCS!

```
[TCOSS] N3/T27 .624=IFX DDI Number=38 OrigNum=131 RedirectingNum=
[TCOSS] N3/T27 .049=Voice Call with CalledId=99, CallerId=131 RedirectingNum=
```

6.5 Message Wait Indication (for the QSIG Protocol Only)

Message Waiting supplementary service makes it possible to control the Message Waiting indicator (MWI) via ISDN FAX module using the new Temporary Signaling Connection (TSC) send mode. The TSC connection is a special type of connection that is dedicated for controlling some features in the PABX system.

It can be activated using following send number syntax:

```
N=<CC>:TC<Command>;[<MsgCenterID>];ExtensionNumber
```

CC channel number

TC send switch for the Temporary Signaling Connection

Command MWION or MWIOFF to switch MWI on/off

MsgCentreID* an identifier for the Message Centre/VoiceMail Server, optional

ExtensionNumber the telephone number where the MWI should be switched on/off

(*) MsgCentreID is per QSIG standard definition an optional parameter but may be mandatory for some PBX implementations. Therefore it may have to be configured properly in the PBX configuration (as for example Hicom 300/Hipath 4000).

6.5.1 How to Use TCOSS MWI Functionality

The most convenient way to use MWI is to create two KCS services for MWI ON and MWI OFF.

Example for service MWION (MsgCentreID is 79699)

```
Address type FREE, prefix F:TCMWION;79699;
```

Example for service MWIOFF

```
Address type FREE, prefix F:TCMWIOFF;79699;
```

Example for service MWION (without MsgCentreID)

```
Address type FREE, prefix F:TCMWION;;
```

Generally there are two possibilities how to use MWI functionality:

1) Using MWI for TCOSS users

Setup Message wait On/Off events to

```
MWION,UserExtension
MWIOFF,UserExtension
```

2) Using MWI for different Mail Platforms (Notes, Exchange, GroupWise)

A specialized MWI MailServer Agent is necessary that will generate MWI control send orders using services MWION/MWIOFF to user's extension.

6.5.2 KCS and PBX MWI Requirements

1. QSIG protocol running via BRI or PRI interfaces, TC20/TC33/TC34 or Model 305 Line Server with TC23/TC24 interfaces (TSxx interfaces are not supported!).
   Take care that proper PBX model is setup in the config line 287, pos. 15 and 16:

   Config line 287, pos. 15..16, default: 00 00 (all PBX models)
   00 01 (Alcatel OmniPCX 4400 R6.x)

2. The PABX system must support the QSIG MWI supplementary service - for detailed requirements please refer to the "PABX Requirements" document and for the list of validated PBX systems the Model 305 PBX Integration Manual

3. The MWI must be configured properly in the PABX configuration

6.5.3 Handling of MWI Error Codes

The MWI send order may fail only in the case of call collision and low-level line problems and breaks with error codes listed in the following table. These are namely the only conditions where it makes sense to perform MWI send order retries:

Error code	ISDN cause	Description	Break code	Possible reason and recommended action
IA	301	Call Collision	1	Call collision with an incoming call. Dedicate one channel for MWI only to avoid it
IJ	202 200	No connection to local exchange (layer 1 or lower layer protocol problem)	2	ISDN line problem (cable disconnected, line deactivated, etc.)

If any other error condition occurs, the MWI send order would be positively terminated anyway as it makes no sense to perform retries if the PBX would report any MWI protocol problem.
So the QSIG MWI send orders are very similar with the "direct dial mode" that is being used by analogue fax interfaces to control MWI lamps by dialing DTMF commands without checking any error code afterwards.

If the MWI send order was positively terminated, the response field of its send order gives the explanation whether it worked or not. Following response fields are defined:

Response text	Possible reason and recommended action
TSC Ok	The MWI command worked fine
TSC Return Result Error	• The PBX does not support MWI QSIG supplementary service (in the case all MWI send orders end up with this response text) • The MWI send parameters (the extension number and/or message centreID) are wrong • The KCS and the PBX are not synchronous concerning MWI status (e.g. KCS tried to set MWION/MWIOFF on an extension which is already MWION/MWIOFF – see more detailed explanation below). This may happen often during normal operation!
TSC Return Reject Error	The QSIG MWI supplementary service is most probably not supported or not properly configured on the PBX side
TSC not supported TSC timeout	The PBX does not support QSIG supplementary services at all as even the temporary signaling connection does not work, perhaps the PBX trunk is not setup for QSIG

MWI ON/OFF events are generated by the email system when the user receives the 1st new message in his mailbox or listens to the last unread message, respectively. It is important that the MWI will be delivered almost immediately to the PABX system. But if one of the MWI send orders fails (e.g. line is disconnected or call collision occurred) it can easily happen that the status of the MWI lamp on the telephone would not be synchronous with the user's inbox. Therefore it does not make much sense to make a lot of retries after a MWI send order hasn't succeeded as the user's mailbox status may change meanwhile. For huge installations with more than 500 users it is a good idea to dedicate one fax module for MWI signalization (switch its fax reception off to avoid call collisions) and change the number of retries for break codes 1 and 2 like following:

Config line 43, new send status after unsuccessful send attempt for BREAK=1

```
'87------0,
```

Config line 44, new send status after unsuccessful send attempt for BREAK=2

```
'87------0,
```


The problems with the MWI status synchronization between user's mailbox and the MWI lamp are not QSIG MWI specific. The same problem may occur if using some proprietary MWI signaling method (via V.24 asynchronous line, via DTMF signaling using analogue FAX module's direct dial mode).

Example of an unavoidable TSC Return Result Error response

TSC Return Result Error response may occur often with MWIOFF send orders for example with Alcatel OmniPCX 4400 PBX in the following way:

1) The user receives a new message and KCS sets his MWI lamp on (MWION send order)

2) The user presses his call back button on the Alcatel phone, the PBX calls KCS server and user listens the message on the phone. The PBX immediately sets the MWI state to OFF

3) After the user has hung off, the KCS server generates MWIOFF send order, it is positively terminated but ends up with the Return Result Error text in the response because the message lamp is already in the off state

This problem could be avoided by NOT generating the MWIOFF send order on KCS at all: but this would work only for those users who access their messages on the Alcatel telephone by pressing the callback button. But it would not work for those users who access their messages at their desktop (e.g. Notes), because in that case the MWIOFF command would be missing.

6.5.4 Testing the MWI Functionality

The easiest way to test the MWI signaling is to use a TCOSS user.

1. Install KCS Server Model 305 and setup one or more UIF modules for QSIG protocol, activate usage of the redirecting number for VOICE calls (UIF config line 295, pos. 3 to 01)
2. Setup the MWI functionality properly on the PABX side
3. Setup TC/VoiceAccess, set the VoiceAccess number (e.g. '9999')
4. Create KCS services MWION and MWIOFF (as described above) with the MsgCentreID parameter set to according the PABX setup (see the table PBX Parameters for MWI)
5. Setup one TCOSS user with a defined voice extension (e.g. '2222'), and activate any type of call forward service (CFU, CFNR or CFB) on the telephone '2222' to the destination '9999'
6. Setup the MWI ON and MWI OFF events for the user '2222' to MWION,2222 and MWIOFF,2222, respectively.
7. Call the extension '2222', let the call be redirected to the KCS server (or call the KCS server directly) and leave a message for the user with extension '2222'

   The MWI lamp on the telephone '2222' must be switched ON: ***If so, MWI ON works OK***

8. Call the VoiceAccess number ‘9999’, login as the user ‘2222’ and listen to the new message(s). Hang up the line.

   The MWI lamp on the telephone ‘2222’ must be switched OFF: ***If so, MWI OFF works OK***

If the test fails it could be that:

KCS MWI implementation is incompatible with the PBX.

The PBX does not support MWI service

Following actions should be performed in the case of problems:

1. Configure TCOSS ISDN trace module before you start any testing with MWI
2. Verify whether the PABX really supports MWI signaling via QSIG protocol and whether it is properly setup on the PABX side, especially the MsgCentreID number
3. Try to use MWI with an empty MsgCentreID (e.g. use the prefix F:TCMWION;; for the MWION service)
4. Save the ISDN trace file and send it to Kofax for evaluation

6.6 Call Transfer (for the QSIG Protocol Only)

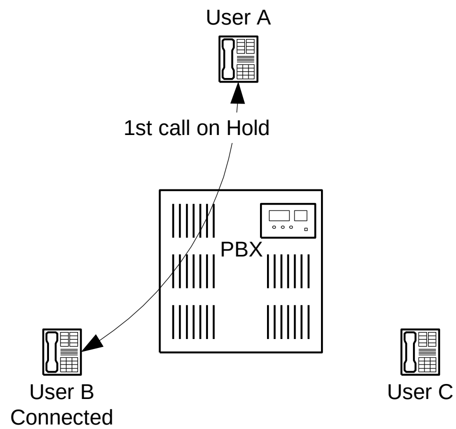
6.6.1 Basic Terms

Per definition Call transfer (CT) is a supplementary service that enables a user A to transform two of his calls (with users B and C) into a new call between users B and C only.

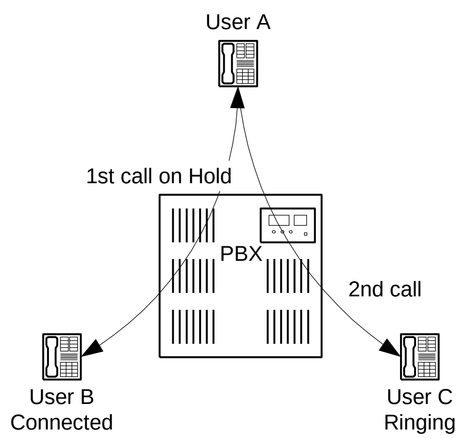
The 1st of the two calls (between users A and B) involved is being referred to as a *primary call* and the 2nd call (between users A and C) as the *secondary call*.

Usually the Call Transfer procedure works like following:

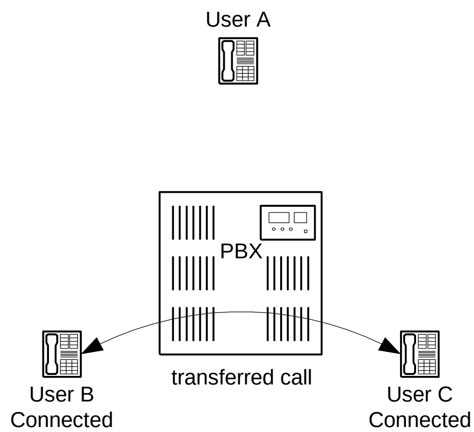
1) User A has a connected call with User B and he puts this call on HOLD (the user B hears the call hold music if provided by the PBX for held calls):



2) User A initiates the secondary call establishment with the user C:



3) User A invokes the Call Transfer procedure and thus transforms two of his calls into a new one between users B and C:



From the procedural point of view, there are three main Call Transfer types:

1) *Blind Call Transfer* (Unsupervised or One-Step Transfer)

   User A invokes Call Transfer immediately after secondary call establishment has been started without verifying the status of User C

2) *Semi-Supervised Call Transfer*

   User A checks only for a busy condition of the User C: if not busy, invokes the Call Transfer immediately

3) *Supervised Call Transfer*

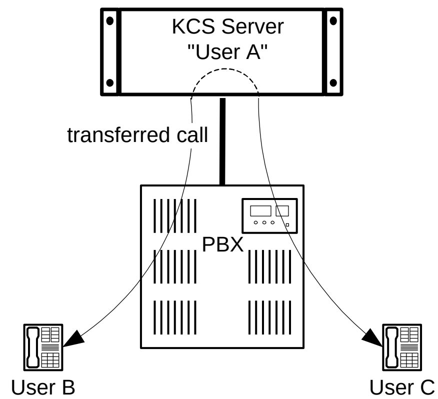
   User A checks for a busy and not responding condition of the User C: it invokes the Call Transfer only if User C answers the call.

   Supervised Transfers are sometimes being referred to as *consulted or announced* Call Transfers – this is if the User A consults the transfer with User C before the Transfer is invoked.

6.6.2 Internal and External Call Transfer

From the switching point of view, KCS server supports two Call Transfer types:

1) KCS server (being the „User A“) switches two of its VOICE calls together using so called *tromboning* technology (A telephony term to indicate the unnecessary double use of a private wire between two telephone systems). The disadvantage of *tromboning* is that two lines towards PABX are occupied for one caller:

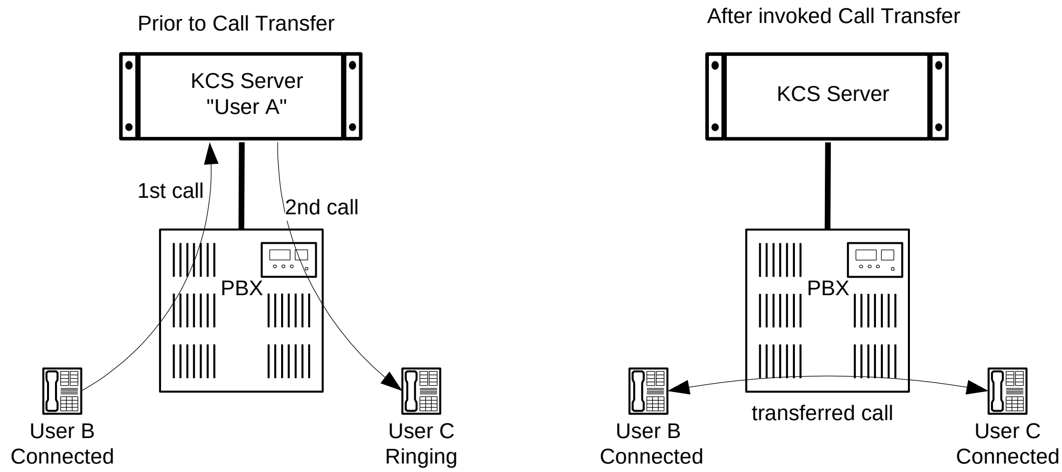


Note that the Call Transfer per “tromboning” does not involve any supplementary services!

Tromboning is also being referred to as *Internal Call Transfer*.

2) KCS server („User A“) uses the Call Transfer supplementary service and in this way it is able to transfer any of its VOICE calls to another destination number via the PABX.

For example, if the user B calls TC/VoiceAccess (“User A”) and decides to be connected further to another destination number (user C), KCS server initiates Call Transfer via the PABX. As a result the PABX interconnects users B and C internally and disconnects both original calls towards KCS server. The advantage is that no lines are occupied between KCS server and the PABX:



Both Call Transfer types supported by the KCS server (tromboning and via Call Transfer supplementary service) can be referred to as *supervised transfers* as they are invoked as soon as User C answers the secondary call.

Usage of Call Transfer supplementary service will be also referred to as *External Call Transfer*.

6.6.3 Two Procedures of QSIG Call Transfer

There are two procedures of QSIG CT supplementary service: *CT by join* and *CT by rerouting*.

With *CT by join* KCS server at first joins (=interconnects) both calls together (and thus still occupies both lines towards PABX). But after the successful *CT by join* the PABX typically triggers the *Path Replacement* supplementary service in order to optimize the route of the new call. As a result, the PABX will interconnect two other users B and C internally and disconnect the calls towards KCS.

There are two possible implementations of Call Transfer by join:

1) *Call Transfer into Alerting*

   As soon as the User A (KCS Server) receives an Alerting notification from the User C (User's C telephone is ringing), he may (optionally) send a Call Transfer indication to both Users B and C in order to notify them about the ongoing Call Transfer and about the new party the call is going to be transferred to: thus User B is being informed the call is going to be transferred to User C and vice versa. If users B and C had telephones with display, they may see the number of the user the call is going to be transferred to on their displays.

   Once the User C answers the call, User A finally joins both calls together, informs the PBX on the invoked Call Transfer and the PBX should trigger Path Replacement in the PBX afterwards.

2) *Call Transfer into Connected*

   User A (KCS Server) simply ignores the Alerting notification from the User C and invokes the Call Transfer as soon as User C answers the call.

   With this variant users B and C are notified about the party the call has been transferred to as late as User C has answered the call.

CT into Connected is a clear disadvantage especially for the User C: as his telephone starts ringing, he sees either no caller number at all or eventually the number of the KCS Server on his display (the number of User B will be provided after he would answer the call). On the other hand, with the CT into Alerting User C sees the number of User B as soon as his telephone starts ringing.

KCS Server supports both Call Transfer into Alerting and into Connected variants as well (per configuration, see below), the former is the preferred method while the latter is provided only for compatibility reasons (as some older PBX SW releases may support only this variant, see Model 305 PBX Integration Manual for details).

<u>The crucial question</u> with Call Transfer is whether PABX is able to trigger Path Replacement after *CT by join*. If not, the *CT by join* brings almost no improvement in comparison with the tromboning, as two lines will be still occupied for one transferred call.

<u>Note that if the PBX triggers Path Replacement after CT by join, two lines are occupied only until User C answers the call.</u>

(QSIG Supplementary Service Path Replacement allows an active call connection through a QSIG private network to be replaced by a new connection after a call route modification like Call Transfer or Call Forward, in order to obtain a more efficient connection. The benefits are reduced costs and increased network efficiency by eliminating unnecessary connections.
KCS Server does neither directly support nor invoke Path Replacement, it is up to the PBX to use it to optimize the connection after the Call Transfer by join).

With *CT by rerouting* KCS server would not have to join both calls together: instead, the *CT by rerouting* procedures require the PABX immediately to make a new connection between users B and C and to disconnect both previous calls towards KCS server.

Note that the CT by rerouting is not supported – neither by the KCS Server nor by the most of known PBX systems. It is mentioned here only for information purposes.

6.6.4 Call Transfer and Call HOLD

Some PABX systems require that the 1st of the two calls involved in the Call Transfer must be put on HOLD before the Call Transfer is being invoked. HOLD usually means that you are listening to the company's hold music while the other call is established.

But this is not required by the CT supplementary service and therefore this option may be configured separately on the KCS side.

6.6.5 Call Transfer Requirements

1) QSIG protocol running via BRI or PRI interfaces, TC16/TC23/TC24 interfaces (KCS Server Model 305)

2) The PABX system must support the QSIG CT by join and Path Replacement supplementary services. The PABX must be able to invoke Path Replacement (route optimization) after successful CT by join. (For detailed PABX requirements please refer to the “PABX Requirements” document and for the PBX configuration to the “Model 305 PBX Integration Manual”)

3) The Call Transfer and invocation of Path Replacement after successful CT by join must be configured in the PABX.

Usage of Call Transfer functionality must be configured within the UIF module by changing the fifth and sixth position of config line 295

:00 00 00 00 00 00 00 00 ,295

Position 6, Call Transfer via Call HOLD function
- 00 …. Call Transfer without Call HOLD
- 01 …. Put the 1st call on HOLD prior to the Call Transfer

Position 5, Call Transfer via PABX
- 00 …. Call Transfer via PABX disabled (tromboning)
- 01 …. Call Transfer by join to Alerting
- 02 …. Call Transfer by join to Connected

6.6.6 Use Cases for the Call Transfer Supplementary Service

Call Transfer is a basis technology for Voice applications that transfer a call to other destination numbers, for example TC/Dialer, TC/Attendant, Call Sender etc.

From the application's perspective, there is one major functional difference between Internal Call Transfer (tromboning) and External Call Transfer (via PABX).

After the call has been transferred externally via the PBX, it does not occupy any lines between KCS and PABX, <u>but the KCS loses the control of this call</u>.

But on the other hand, after the call has been transferred locally via tromboning, it occupies two lines between KCS and PABX, <u>but the KCS Server keeps control of this call</u>.

It is clear that both scenarios have their advantages and disadvantages and applications themselves should decide which Call Transfer method to use.

Therefore KCS VOICE applications to initiate Call Transfer specify their preferred Call Transfer method – external or internal:

1) If the application requests an Internal Call Transfer, KCS Server always do so (regardless of the Call Transfer method configured in the UIF module)

   The only application that by default requests an Internal call Transfer, is the Call Sender function of TC/VoiceAccess - for the simple reason: after the user who is just listening to his voice mails has elected to call one of his voicemail senders, he can go on accessing his voice mails after this called person has hung up the line.
   (Having used an External Transfer, the user would have to call KCS Server again to go on).

2) If the application requests an External Call Transfer, KCS Server does so if Call Transfer via PBX is configured: if not, it defaults to tromboning.

6.6.7 Testing the Call Transfer by Join

The crucial problem with Call Transfer by join is whether the PABX initiates the Path replacement (PR) after the successful Call Transfer by join or not. If it wouldn't, even with successfully performed Call Transfer by join two lines between KCS and PABX would be still occupied and thus the Call Transfer would not bring any improvement in comparison with tromboning.

The easiest way to test the **Call Transfer with Path replacement** is to use the TC/VoiceAttendant functionality (see TC/VoiceAccess documentation).

1. Install KCS Server Model 305 and setup one or more UIF modules for QSIG protocol, setup CT by join + Call HOLD in the UIF config line 295. Additionally setup an ISDN Tracer Module
2. Setup TC/VoiceAccess with TC/VoiceAttendant, set the Transfer Number to any internal telephone extension available (e.g. '1111') and the VoiceAccess number (e.g. '9999')
3. Setup one TCOSS user with a defined voice extension (e.g. '2222')
4. Call the extension '2222' from another internal phone (either directly or via call redirection of the telephone '2222'). You will hear the welcome prompt for the user '2222' and press '0' to be connected to an operator.
5. The telephone '1111' starts ringing and you should hear the PABX HOLD music and may observe 2 active calls with TCMON
6. Answer the call on the telephone '1111'.
7. Now you should be connected with the telephone '1111' and after a few seconds both active calls should have disappeared from TCMON channel view. If so and you are still connected with tel. '1111', **everything works OK**. If NOT, please see possible problems below.

Possible problems might be:

1. You are connected with tel. '1111', but TCMON still indicates 2 active connections

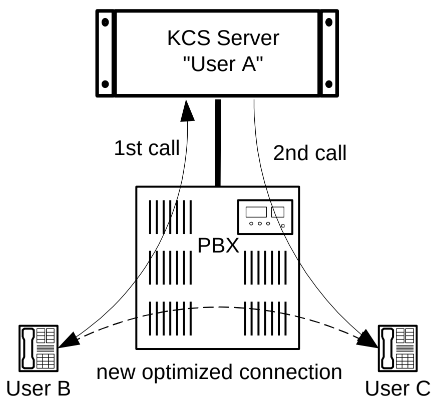
   The CT itself has worked but the PABX has not activated PR (route optimization) afterwards
2. After you have pressed '0' for operator, your call was disconnected and also TCMON hasn't shown any active calls

   There may be a more difficult CT-related protocol problem between KCS and the PABX. Perhaps different protocol is configured on the PABX side or there is QSIG but not properly setup for Call Transfer and Path replacement
3. After you have pressed '0' for operator, you haven't heard the PBX HOLD music

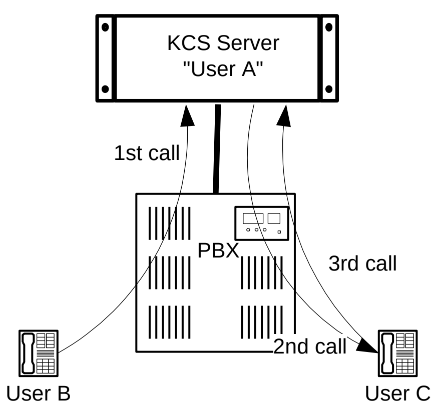
   switch call HOLD off (UIF config line 295) and try again

6.6.8 Call Transfer Connection Loop Detection

If KCS Server (“User A”) transfers one of its calls (with User B) to other destination (User C), at first it makes a 2nd call to User C. At the moment the 2nd call gets answered, KCS Server invokes the Call Transfer and the PBX optimizes the connection loop (via the KCS Server) by triggering the Path Replacement supplementary service: the PBX recognizes the loop, interconnects the User B and C via the new optimized connection and disconnects both 1st and 2nd calls towards KCS server:



But the problem is that the PBX is able to optimize the call route only after the 2nd call has been answered. If the 2nd call would not be answered but would be forwarded back to the KCS Voice Server (due to call forward on busy, not responding or unconditional), the 3rd call towards KCS server would be established and the User B would be connected to the User’s C mailbox while effectively occupying 3 lines towards KCS server. And the situation may easily escalate if the User B would decide to be connected to another extension. In this way, one caller may easily occupy 3, 5, 7 or even more lines between KCS server and the PBX:



In order to prevent such a looping the Call Transfer loop detection mechanism may be used.

Its principle is quite simple: after the KCS server has recognized that the PBX attempts to forward the 2nd call back to the own line (which would result in connecting the User B with User C's mailbox via 3 lines), it disconnects the 2nd call and switches the User B internally:

1) Either to the User C's mailbox (default behavior)

2) Or to the User A's mailbox (while providing an information to User B that User C could not be reached)

Which of these two methods will be used is the Voice application's configuration option, see TC/VoiceAccess manual.

6.6.8.1 Prerequisites

1) QSIG protocol running via BRI or PRI interfaces, TC16/TC23/TC24 interfaces (KCS Server Model 305)

2) The PBX system must support the QSIG Call diversion supplementary services (for a list of PBX devices that support this requirement please refer to the "Model 305 PBX Integration Manual")

3) The PBX system should support the QSIG CT by join and Path Replacement (PR) supplementary services. The PBX must be able to invoke Path Replacement (route optimization) after successful CT by join (this is necessary for the PBX to optimize the call route after the 2nd call has been answered)

4) All telephone extensions must be forwarded to the same unique KCS Voice Access number (this is necessary to recognize the loop on the KCS side)

5) TC/VoiceAccess version 3.02.02 or later

6.6.8.2 Configuration

The KCS Voice Access number must be configured with the line type 'L' in the UIF number conversion facility (config lines 254-283):

```
'Laaaa=Vaaaa                  ,254 aaaa is the Voice access number
```

For the TC/VoiceAccess configuration please refer to the TC/VoiceAccess documentation.

6.6.8.3 How to Test Call Transfer Loop Detection

The easiest way to test the loop detection is to use the TC/Attendant function (see TC/VoiceAccess documentation).

1) Install KCS Server Model 305 and setup one or more UIF modules for QSIG protocol

2) Setup TC/VoiceAccess with TC/Attendant, set the TransferNumber to any internal telephone extension available (e.g. '1111') and the VoiceAccess number (e.g. '9999')

3) Set the TC/VoiceAccess registry key "AttendantLoopDetection" to "SBOX"
(this means that the caller should be switched to operator's mailbox after the Attendant loop has been detected)

4) Configure the call forward on the telephone ‘1111’ to Voice access number ‘9999’

5) Setup one TCOSS user with defined voice extension (e.g. ‘2222’)

6) Setup the Voice Access number in the UIF number conversion facility:
```
'L9999=9999    ,254 Voice Access number for loop detection
```

7) Call the extension ‘2222’ from any other phone (either directly or via call redirection of the telephone ‘2222’). You will hear the welcome prompt for the user ‘2222’ and press ‘0’ to be connected to an operator.

8) The telephone ‘1111’ starts ringing, don’t answer the call but let it be forwarded (due to not responding) to the KCS server.

9) Now you should be connected with user’s ‘1111’ mailbox and only one active call should be indicated in the TCMON: if so, everything works OK.
If NOT (you see 3 calls in the TCMON), please see Possible problems below.

Possible problems:

1) The PBX does not support QSIG Call Diversion supplementary services or a different protocol is being used (not QSIG)

2) KCS Server has not recognized the loop, due to:
   - TC/VoiceAccess version older than 3.xx.xx is being used
   - The Voice access number was not setup with L-line type in the numb. conv. facility
   - The telephone ‘1111’ was not forwarded to the KCS Voice Access number

3) There is a different problem: please make an ISDN trace of this test procedure and contact TCINT.

6.6.9 Restrictions with the Call Transfer Functionality

The following restrictions/problems are known so far:

1) Many PBX systems seem to have a restriction that if the switchboard (operator) is involved in one of the calls to be transferred, the Path Replacement will not be triggered after Call Transfer by join. As a consequence, 2 lines per transferred call would still remain occupied.
Refer to the Model 305 PBX Integration manual for details.

7. KCS Primary Rate Configuration

The next chapter describes the procedure of configuring and implementing the KCS primary rate ISDN solution at the customer. Only primary rate is discussed in detail as it is usually more complicated and offers several possibilities to be configured. The following description should help you in configuring and understanding the KCS ISDN primary rate solution.

7.1 Hardware Requirements

Hardware requirements are not described in detail within this description here as they're fully covered within the KCS hardware documentation. Below a short summary of things to take care of while installing primary rate from the hardware point of view:

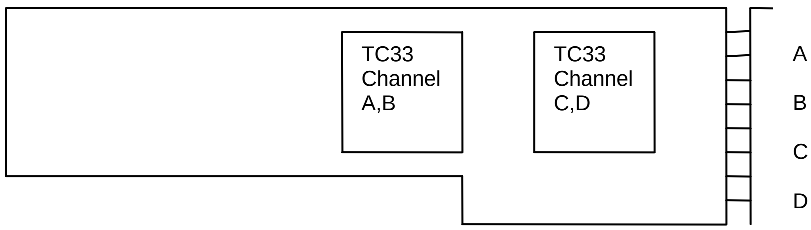
- TC20 and TC33 interfaces are required with correct revision and EPLD number
- Jumper on the TC33 interfaces has to be set correctly
- Termination settings has to be done correctly
- Line termination has to be done correctly (75 or 120 Ohm)
- TP80 has to be jumpered correctly
- TC81 has to be jumpered correctly

Please note that there are more things to take care of which are described in detail in the relating hardware documentation. Below a quick list of things to take care of:

- One TP80 can handle a maximum of 6 TC20 interfaces
- One TC20 can have two TC33's connected to (so one TP80 can handle 24 lines)
- A TP80 has 4 different hardware addresses which influences the configuration, specially the slave numbering. A hardware address of 150 HEX relates to slave 1.1, 140 HEX to 1.2, 160 HEX to 1.3 and 170 HEX to 1.4
- One TC20, which holds two TC33 interfaces uses channel numbering A,B,C and D
- One TC20 which holds the TC34 and one TC33 can NOT use channel C and D (used by TC34)
- The TC34 is NOT configured (is only a hardware component)

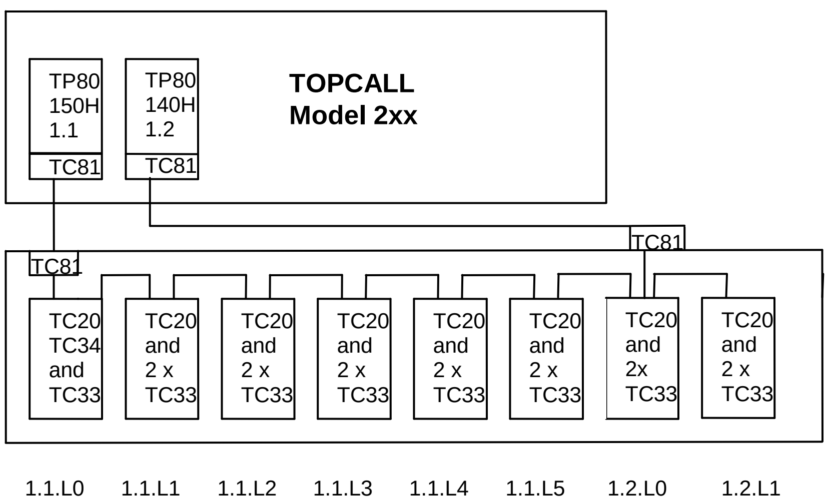
7.1.1 TC20 Connection Ability

The following drawing shows a quick overview about TC20's connection ability



As the TC34 is connected at the rear of the TC20, it is now clear why channels C and D can't be used anymore.

A full equipped E1 primary rate single KCS might look like in the drawing below:



7.2 KCS ISDN's Reference Channel

To simplify creation and maintenance of UIF channels the following features exist:

When creating a new UIF channel you will be prompted for a reference channel (after selecting slave/slot). The input is stored in config line 249. If the field is set to blanks, no reference channel is used. This means that WCONFIG behaves as with prior releases.

The reference channel itself can have config line 249 set to blank or can contain it's own channel number.

A reference channel is specified by its channels number (e.g. 04). It must share the same physical ISDN line with the new UIF. The configuration of the reference channel is used instead of the standard configuration.

The WCONFIG program remembers the last entered value and offers the same value as default next time. Thus, when creating a lot of UIF channels for a primary rate ISDN line this is simply a matter of pressing OK there.

After doing a “Single Channel” configuration of a primary rate ISDN reference channel you will be asked if you want to copy this configuration to all channels that belong to this reference channel or if you want to copy only ISDN specific details of the configuration (below line 249) to all channels that belong to this reference channel.

```
Copy complete to all channels which use this channel as reference
Only copy below line 249 (line 250,251, .. end)
Do not copy anything
```

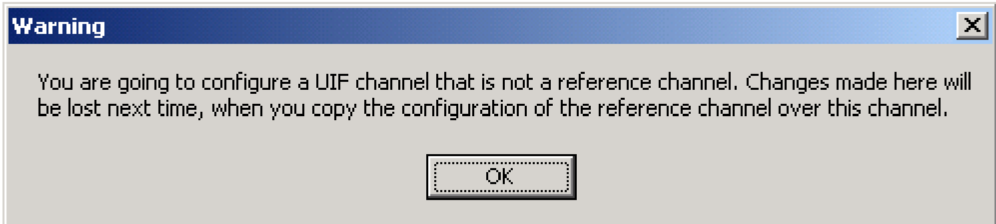
Please keep in mind that unlike a “Configure all Channels” functionality this will copy the entire (or the configuration below 249, respectively) to all the channels and not only the changes made! There may be only one reference channel for a group of channels sharing the same physical line and it must always be the one with the lowest channel number.

Use of reference channel is optional with basic rate ISDN and mandatory for primary rate ISDN configuration.

7.3 Software Configuration Rules

1. The TC34 interface can be used in position 1 (near back plane) of TC20 only. It must be used together with a TC33 interface in position 0 (near processor) of the same TC20. This TC20 should be used in slot L0 only!
2. The TC34 cannot be explicitly configured. Its slot/position is implicitly configured by the position of the reference channel. Therefore the reference channel must always be configured on that TC20 with TC34 interface.
3. Start configuration with the first UIF on channel A of the TC20 interface used for TC34. This is the reference channel of primary rate line.
4. Configure UIF modules for all other TC33 interfaces connected to the primary rate line. The channel number of the reference channel (configured before) must be specified. These channels must have a higher channel number than used for the reference channel.
5. For optimum distribution of processor performance on the TC20 interfaces, it is recommended to start configuring all a-slots, following by all b-slots, c-slot and d-slots as shown in the example below.
6. Configure the reference channel with “Configure - Single Channel”. Refer to next chapter for configuration details. Upon exit, you will be asked if you want to copy this configuration to all channels that belong to this reference channel or if you want to copy only ISDN specific details of the configuration (below line 249) to all channels that belong to this reference channel.

Unlike a “Configure all Channels” this will copy the entire (or the configuration below 249, respectively) to all the channels and not only the changes made! If attempting to perform configuration changes on a non-reference primary rate channel the Config program displays the warning



but despite of this you may configure all config lines except those below line 249 on a non-reference primary rate channel (e.g. setting different channel group on a sub-group of channels etc.). ISDN config part (lines below line 249) **must be identical** for all channels sharing the same physical line.

First example of a Primary rate configuration (not all channels listed)

Channel	SW-Modul	Speed	HW-Modul	Slot	Slave
04)	IF		TC20-A	L0	1.1
05)	IF		TC20-A	L1	1.1
06)	IF		TC20-A	L0	1.2
07)	IF		TC20-A	L1	1.2
08)	IF		TC20-A	L2	1.2
09)	IF		TC20-A	L3	1.2
10)	IF		TC20-A	L4	1.2
11)	IF		TC20-A	L5	1.2

According to configuration rule number 5, all A-channels have been configured first. After all A-channels all B-channels will be setup, followed by the C and D channels. Channel number 04 is the reference channel and used for primary rate specific settings.

This is only a configuration suggestion. In the next example the same configuration is done with "standard" slave numbering.

Second example of a Primary rate configuration (not all channels listed)

Channel	SW-Modul	Speed	HW-Modul	Slot	Slave
04)	IF		TC20-A	L0	1.1
05)	IF		TC20-B	L0	1.1
06)	IF		TC20-A	L1	1.1
07)	IF		TC20-B	L1	1.1
08)	IF		TC20-C	L1	1.1
09)	IF		TC20-D	L1	1.1
10)	IF		TC20-A	L0	1.2
11)	IF		TC20-B	L0	1.2
12)	IF		TC20-C	L0	1.2
13)	IF		TC20-D	L0	1.2

This is the second possibility to configure primary rate. You see clearly that the TC20-C and D definition for slave/slot 1.1.L0 does not exist as this is the TC34 interface which is not configured!

In both examples, not all ISDN channels have been listed. For E1 connections you should configure 30 channels while for T1 connections only 23 channels are in use.

After you've set up the complete hardware/software assignment, you must define the primary rate specific ISDN settings. The most important line for setting up a primary rate connection is config line 250, see below:

:01 00 FF 00 00 ,250

Position 5, ISDN Line Coding

00 …. A-Law coding (Europe)
01 …. μ-Law coding (North America, Japan)

Position 4, ISDN line type

00 …. Basic rate ISDN connection (default value)
01 …. Primary rate E1 ISDN connection (Europe)
02 …. Primary rate T1 ISDN connection (North-America, Japan)

Position 3, TEI assignment procedure

00 …. Used for Point to Point connections
01..3F non-automatic TEI value
FF ….automatic TEI assignment used (for PTMP)

Position 2, Layer 3 protocol options

00 …. Euro ISDN or Q.931 protocol
01 …. 1TR6 protocol (old German national)
02 …. AT&T protocol for 4ESS switch type
03 …. QSIG protocol

Position 1, Connection Type

00 …. Point to Point
01 …. Point to Multipoint

Please note that a Primary rate ISDN connection is always a Point to Point connection, while a Basic rate ISDN connection can be defined as “Point to Point” or “Point to Multipoint” connection. For E1 connections this line is therefore changed to :00 00 00 01 00, while for T1 connections, the line is defined as :00 02 00 02 01.

7.4 Outgoing B-Channel Allocation

With every ISDN access there is a pool of logical B-channels (bearer-channels) that may be used for Voice or FAX transmission. The standard number of B-channels used depends on the type of ISDN access

- Basic rate interface offers two B-channels B1 and B2
- Primary rate E1 offers 30 B-channels B1, B2, … ,B29 and B30
- Primary rate T1 offers 23 channels B1, B2, …,B22 and B23.

The B-channel used for specific FAX transmission is being negotiated between the user side (KCS) and the network side (PABX or PTT) during the call establishment phase. Each of the sides may request:

Any of the available B-channels from its peer side

This is the simplest negotiation method and is being used with basic rate interface: the B-channel allocation and assignment is completely done by the network side. There is no risk of B-channel assignment collision. Please keep in mind that this method is supported ONLY with Euro-ISDN and 1TR6 protocols running via basic rate interface.

Specific B-channel 'preferred' from its peer side and any alternative channel acceptable

The call originator requests specific B-channel that is currently not used by another call, e.g. B10. If the requested channel is free also at the peer side, it allocates it for this call. If the requested channel has already been allocated at the peer side, it allocates any other free B-channel available for this call. If there is no free B-channel available at the peer side, the call is cleared.

Specific B-channel 'exclusive' from its peer side and an alternative channel NOT acceptable

The call-originator requests specific B-channel that is currently not used by another call, e.g. B10. If the requested channel is free also at the peer side, it allocates it for this call. If the requested channel has already been allocated at the peer side, the peer side clears this call as the channel is not free.

7.4.1 B-channel Assignment Collision Problem

When both sides at the user-network (KCS and PABX, PTT) interface request the same B-channel at the same time, the B-channel assignment collision occur. In a worst case, e.g. if both sides have requested the same B- channel as 'exclusive', both calls may be cleared even though a couple of other B-channels may be free.

In order to avoid unnecessary collisions and subsequent call clearings, channel-selection parameters "**exclusive – no alternative available**" or "**preferred – any alternative acceptable**" (ISDN config line 286, pos. 3) and "**A-side**" or "**B-side**" (ISDN config line 286, pos. 4) should be properly configured.

:00 00 00 00 00 00 00 00 ,286

Position 4, Call collision handling

00 …. A-side, The TC34 interface starts selecting a B-channel for outgoing calls from the lower end B-channels, B1, B2…..B23(B30)
01 …. B-side, The TC34 interface starts selecting a B-channel for outgoing calls from the upper end B-channels, B23(B30) …B2, B1

Position 3, Outgoing B-channel negotiation method

00 …. Any channel selected (only for Basic rate interface except for QSIG Protocol
01 …. Exclusive – no alternative available, for Primary rate ISDN
02 …. Preferred – Any alternative accepted, for Primary rate ISDN

The "Preferred – any alternative accepted" selection method is recommended for KCS configuration as it is not so restrictive like "Exclusive – no alternative available". The network side may allocate any other channel which is currently free instead of clearing the call due to the requested channel busy status.

The “A or B-side” parameter states the direction of selecting the requested B-channel for an outgoing call. The “A-side” of the interface always starts selecting a B-channel from the lower end (B1, B2, …) and the “B-side” from the upper end (B30, B29, …). Additionally, the “A- side” of the interface should be awarded the channel with higher priority during “symmetrical” requests for the same channel at the same time.

Symmetrical means that both A and B-side uses either “preferred” or “exclusive”, while **non-symmetrical** means that A or B side uses preferred, while the other side uses the exclusive channel negotiation method.

A-side defined as Preferred and B-side defined as Preferred

In this case, the B-channel should be awarded to the A-side of the interface, the B-side should be awarded to any other free B-channel which is available

A-side defined as Exclusive and B-side defined as Exclusive

The B-channel should be awarded to the A-side of the interface and the call request from the B-side should be cleared.

During “**non-symmetrical**” B-channel requests at the same time, the “exclusive” side should always be awarded the B-channel. In this scenario, exclusive overrides the A-side priority.

A-side Exclusive and B-side Preferred or A-side Preferred and B-side Exclusive

The B-channel should be awarded to the side which requested the channel as “Exclusive”. The other side should be awarded to any other free channel which is available.

In order to minimize the risk of channel assignment collision problems, both sides of the interface should have both parameters set differently. The most appropriate configuration is network side “exclusive”, A-side and user side (KCS) “preferred”, B – side, as it handles incoming calls from the network with higher priority.

Independent on the network settings, the KCS configuration should handle incoming calls with higher priority. Therefore it is recommended to define KCS as B-side, Preferred. Please note that this is only a recommendation and other configuration settings might work too!

7.4.2 Fractional B-channel ISDN lines

Primary rate ISDN interface which is running via standard E1/T1 lines supports 30/23 B-channels. But at several customer installations also restricted E1/T1 lines are very common. They provide only a limited number of active B-channels (e.g. 10, B1-B10).

There may be several reasons for restricting the number of B-channels. Either the customer does not need the full E1/T1 line (less message traffic) and it is cheaper to order one restricted E1/T1 line as to order several distinct a/b lines. The same E1/T1 line may be shared by different applications (e.g. T1 channels B1-B10 for telephone calls and B11-B23 for FAX calls).

Such a restricted E1/T1 line is being referred to as “fractional E1/T1 line”. Working with such a restricted E1/T1 line for KCS two important things should be kept in mind:

1. The maximum number of active B-channels

2. Which specific B-channels may be used (e.g. B1…B10 or B10…B23) or whether only the number of concurrently used B-channels is limited (e.g. maximum 10 B-channels of any E1/T1 line available)

There is a possibility to explicitly configure B-channels used via the E1/T1 line by the config line 287, pos. 1 – 4. By default all 4 of these config positions are set to 00, which means that all standard B-channels are available via E1/T1 lines. Please note that 00 00 00 00 has the same meaning as 7F FF 7F FF – both can be used to define a full primary rate ISDN connection.

This may be the proper and the easiest configuration for fractional E1/T1 lines where only the number of concurrently used B-channels is restricted, but with no specific B-channel restriction. If there is also a restriction on using specific B-channels (e.g. only B1-B10), B-channels allowed must be configured via config line 287, position 1 up to 4.

Requested B-channels are configured bitwise according to the following drawing below. E1 and T1 B-channels follow a different configuration scheme: there are separate E1/T1 configuration tables. Both tables provide also some examples on configuring specific B-channel range.

The E1 B-channel configuration line

$$:\underbrace{00\ 00\ 00\ 00}_{}\ 00\ 00\ 00\ 00\ 00,287$$

Position 1…4, Allocation of B-channels

00 00 00 00	full ISDN lines available
7F FF 7F FF	same meaning as above, alternative option for all 30

The T1 B-channel configuration line

$$:\underbrace{00\ 00\ 00\ 00}_{}\ 00\ 00\ 00\ 00\ 00,287$$

Position 1…4, Allocation of B-channels, Position 4 not used as only 23 channels available

00 00 00 00	full ISDN lines available
7F FFFF 00	same meaning as above, alternative option for all 23 channels

<u>Configuring the fractional E1 B-channels</u>

Things to take care of / to keep in mind.

- The default value for all ISDN channels is 00 00 00 00 which is the same as 7F FF 7F FF
- Calculation of B-channels is done by adding the decimal bitvalues of the used B-channels as displayed within the drawing below. The resulting decimal value has to be converted to a HEX value and must be defined in the correlating configuration position.
- Calculation of B-channels can be done alternatively with binary values. In this case, a used B-channel has a binary 1, while an unused B-channel has a binary 0

Bit →

7	6	5	4	3	2	1	0

Bit 7 Decimal Bitvalues Bit 0

128	64	32	16	8	4	2	1

1st position 00
B-channel 1-7

E1 B-channel number

NA	1	2	3	4	5	6	7

NA .... Not assigned, BIT value is always 0

Bit

7	6	5	4	3	2	1	0

2nd position 00
B-channel 8-15

E1 B-channel number

8	9	10	11	12	13	14	15

Bit

7	6	5	4	3	2	1	0

3rd position 00
B-channel 16-22

E1 B-channel number

NA	16	17	18	19	20	21	22

Bit

7	6	5	4	3	2	1	0

4th position 00
B-channel 23-30

E1 B-channel number

23	24	25	26	27	28	29	30

Example: Customer wants to use B-channels 6 to 14 for TOPCALL FAX send orders via primary rate

The first config position covers B-channel 6 and 7, therefore Bit 0 and 1 are used. The Bitvalue for Bit 0 is 1 and for Bit 1, 2. Add these values, 1+2 = 3, convert this decimal value to HEX = 03 HEX.

The second config position covers B-channel 8 up to 14, therefore, Bits 7, 6, 5, 4, 3, 2 and 1 are used. The Bitvalue For Bit7 is 128, for Bit6 is 64, for Bit5 is 32, for Bit4 is 16, for Bit3 is 8, for Bit 2 is 4 and for Bit 1 is 2. Add these values, 128+64+32+16+8+4+2=254, convert this decimal value to HEX= FE HEX.

By using channels 6 up to 14, config line 287 has to be set to **03 FE 00 00**

Configuring the fractional T1 B-channels

Things to take care of / to keep in mind.

- The default value for all ISDN channels is 00 00 00 00 which is the same as 7F FF FF 00
- Calculation of B-channels is done by adding the decimal bitvalues of the used B-channels as displayed within the drawing below. The resulting decimal value has to be converted to a HEX value and must be defined in the correlating configuration position.

- Calculation of B-channels can be done alternatively with binary values. In this case, a used B-channel has a binary 1, while an unused B-channel has a binary 0

Bit →

7	6	5	4	3	2	1	0

Bit 7 Decimal Bitvalues Bit 0

128	64	32	16	8	4	2	1

1st position 00
B-channel 1-7

T1 B-channel number

NA	1	2	3	4	5	6	7

NA .... Not assigned, BIT value is always 0

Bit

7	6	5	4	3	2	1	0

2nd position 00
B-channel 8-15

T1 B-channel number

8	9	10	11	12	13	14	15

Bit

7	6	5	4	3	2	1	0

3rd position 00
B-channel 16-23

T1 B-channel number

16	17	18	19	20	21	22	23

4th position 00

Not used for T1 as only 23 channels are available!

Example: Customer wants to use B-channels 16 to 23 for TOPCALL FAX send orders via primary rate

The third config position covers B-channel 16 up to 23, therefore, Bits 7, 6, 5, 4, 3, 2, 1 and 0 are used. The Bitvalue For Bit7 is 128, for Bit6 is 64, for Bit5 is 32, for Bit4 is 16, for Bit3 is 8, for Bit 2 is 4, for Bit 1 is 2 and for Bit0 is 1. Add these values, 128+64+32+16+8+4+2+1=255, convert this decimal value to HEX= FF HEX.

Alternatively, the Binary calculation method may be used. We know that a channel in use has binary 1, while a channel not used has binary 0. Config Position 1, channels 1 to 7 are not used, therefore binary 0000 0000 is calculated. Config position 2, channels 8 to 15 are not used, therefore binary 0000 0000 is calculated. Config position 3, channels 16 to 23 are used, therefore binary 1111 1111 is calculated, in total 0000 0000 0000 0000 1111 1111. Convert this to HEX and the result will be 00 00 FF.

By using channels 16 up to 23, config line 287 has to be set to **00 00 FF 00**

7.5 Special AT&T (4ESS) Primary Rate Settings

The AT&T US primary rate offers special ISDN settings which are described in the following chapters. Please note that they are only valid for AT&T integration and are not used in any other case!

7.5.1 AT&T Service Type

AT&T offers some special ISDN services which can be used during outgoing call establishments. These services are called, SDN (software defined network), MEGACOM and Long Distance Service. These services are defined within ISDN config line 286, position 7.

:00 00 00 00 00 00 00 00 ,286

↓

Position 7, AT&T service used for outgoing calls

00 …. No specific service used for the call on the "call by call" basis, must be provisioned for the whole T1 trunk
01 …. SDN service – Software defined network
02 …. Reserved
03 …. MEGACOM service
04 …. Reserved
05 …. Reserved
06 …. Reserved
07 …. Long Distance Service

7.5.2 AT&T National and International Definition

AT&T ISDN network must be told during call establishment, whether the dialed number is national or international. This task is accomplished by config lines 254 up to 283 (so called number conversion lines): The first entry removes the international prefix "011" commonly used in the USA and inserts the prefix TI instead. The second entry inserts the prefix "TN" for all other – national numbers.

The prefixes "TI" or "TN" instruct the ISDN module to mark the dialed number to be either international or national. The prefixes themselves are removed before dialing. Would any customer like to use different international prefix instead of "011" (e.g. "00"), please change the config line 254 like in the following example

Standard configuration			Change to following configuration		
`'8011~,TI~`	,	254	`'800~,TI~`	,	254
`'8~,TN~`	,	255	`'8~,TN~`	,	255

7.5.3 AT&T Layer 1 Cable Length

By default, the primary rate ISDN layer 1 for T1 installations is set-up to support a cable length between 0 – 35 meter to the NT. Dependent on the type and length of the cabling it may be necessary to increase the output levels. To increase the levels position 3, 4, 5 and 6 in ISDN config line 291 must be changed equally.

The default value is 00 00 00 00 which is internally handled by TCOSS as a decimal value of 20 (HEX 14) which causes an output level of 3 Volts peak according to ANSI T1.408. These values could be increased up to decimal 31 (HEX 1F).

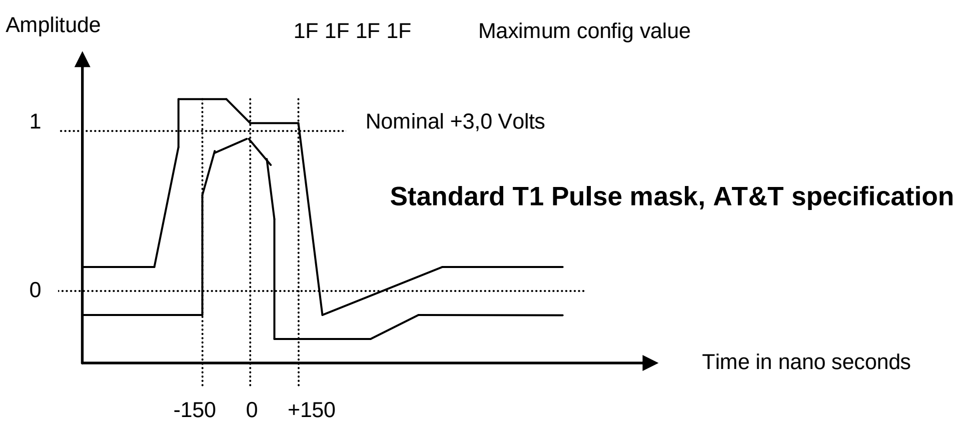
Important to know is that the levels must not exceed 3.6 Volts peak at the receiving side. A level which is too high may cause line crosstalk, which produces excessive errors. In general, a high amount of CRC errors or code violations indicates a wrong output level.

$$:00\ 00\ \underbrace{00\ 00\ 00\ 00}\ 00\ 00\ 00\ 00\ 00\ 00\ 00\ 00\ 00\ 00,291$$

Position 3..6, T1 transmit pulse shape (quarters of full wave, therefore 4 positions)

Please note: Eventhough TCOSS offers higher values, the maximum value should be 3,6 Volts peak (18 18 18 18) as otherwise line crosstalk occurs

00 00 00 00	Default pulse shape value for cable length of 0 up to 35 meters This value is interpreted by TCOSS internally as 14 14 14 14 or decimal 20 which causes an output level of 3 Volts peak.
15 15 15 15	Output level of 3,150 Volts (an increase of 150 milli volts)
16 16 16 16	Output level of 3,300 Volts (an increase of 150 milli volts)
1F 1F 1F 1F	Maximum config value



7.6 Primary Rate E1/T1 Line Status

The primary rate ISDN lines for E1/T1 offers some status information which defines signaling between the network side and the user side (KCS). They are described in the following chapter

7.6.1 Line status signals

ISDN primary rate standards define the following signals between the network and the user side:

- Normal operational frames
- RAI - remote alarm indication
- LOS/LOF – loss of signal / loss of frame alignment
- AIS – alarm indication signal

These status signals may be accompanied by CRC error information.

Two LED’s on the TC34’s back plane gives currently the only information on the E1/T1 line status. The behavior with E1 and T1 lines is basically the same. The only difference is that with E1 lines the line status change almost immediately leads to TC34 LED status change, while with T1 lines, it may last up to 10 seconds until LED’s change their status after line status has changed.

7.6.2 Line status description

The following shows a short description of the status signal which may occur on a primary rate ISDN E1/T1 line.

Normal operational frames, OK condition on the line, no errors detected

RAI – remote alarm indication, also called “**yellow alarm**”, occurs if the user side or network side detects loss of layer 1 capability towards itself, it sends a RAI to the peer side. In case the TC34 interface sends a RAI signal or is receiving a RAI signal, the red LED is on!

LOS/LOF – Loss of signal / Loss of frame alignment, is usually generated if no pulse transitions have occurred for a certain time period. Indicates a bad receiver signal quality on either user side or network side. Whenever the receiver detects LOS or LOF, it responds by sending a RAI signal to the peer side. In case the TC34 interface receives the LOS signal, the green LED is off. If no LOS is received, the green LED is on.

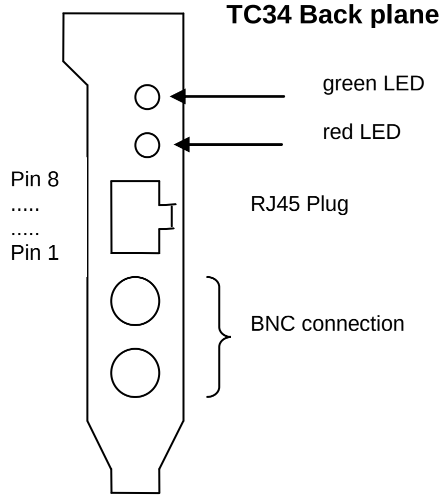
AIS – alarm indication signal, also called “**blue alarm**”. It is sent only from the network side to the user side (KCS) to indicate the loss of layer 1 capability in the “network to user” direction on the network side of the network to user interface. It indicates a network problem. The user side responds to the AIS signal with a RAI signal. In case the TC34 interface receives the AIS signal, the red LED is on

CRC error information, If the receiver, either user side or network side, detects CRC errors within received ISDN frames, it sends the CRC error information to the peer side.

7.6.3 Verifying the E1/T1 Line Status with TC34 LED’s

The TC34 interface which is used for primary rate connections only, offers two LED’s on the back plane. One green LED and one red LED, see TC34 back plane below:

TC34 Back plane



<u>Case 1 – only the green LED is on</u>

This is the “normal” status and means everything ok, ISDN Line is operational, Synchronization between KCS and the PABX or PTT was successful, framing ok, line coding ok.

<u>Case 2 – only the red LED is on</u>

In this case, the TC34 interface receives a LOS – loss of signal (green LED is off) which means that there is obviously no signal on the line. The TC34 in this case sends a RAI – remote alarm indication to the network (as response to LOS). **Starting TCOSS with a primary rate connection without cable attached to the TC34 leads also to a “only red LED on” behavior.** The recommended actions in this case are:

- Verify the E1/T1 cable - TC34 pins 4 and 5 are KCS receiver pins and must be connected to networks transmitter pins. TC34 pins 3 and 6 are KCS transmitter pins and must be connected to networks receiver pins.
- Check the PIN assignment
- Check the cable length (should not exceed 250 meters or 820 foot)

Case 3 – both red and green LED’s are on

In this case, the TC34 interface does not receive a LOS – loss of signal (green LED is on). Additionally the red LED is on which might have the following meanings:

- TC34 sends a RAI – remote alarm indication as response to a LOF – loss of frame condition. The cause might be framing or synchronization problems.
- TC34 sends RAI – remote alarm indication as response to AIS – alarm indication signal. The cause might be the network which sends the AIS signal
- TC34 receives RAI – remote alarm indication from the network as the network detects a LOS or LOF condition from the TC34.

The recommended actions in this case are:

Verify the correct line coding and line framing defined within ISDN config line 291 (CRC-4, double frame, ESF framing, superframing….).

Verify the output signal level (only for T1 installations) via config line 291. Increase the output level step by step – as described!

Case 4 – no LED is on

In this case, the TC34 interface is simply not initialized correctly or damaged. Before replacing the TC34 check ISDN config line 250, position 4 which defines whether the ISDN line is a basic rate or primary rate line. In most cases, this ISDN config line is wrong!

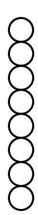
Case 5 – network reports CRC errors or code violations

This may also indicate a wrong output signal level, line length problem or a slightly damaged cable. Before doing any hardware changes, replace the cable with a complete new one.

7.6.4 Verifying the TC34 with a Local Loopback Test

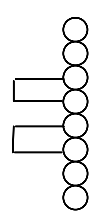
You need for this test a RJ45 plug with a short cable attached (10 cm are enough), where the pins 3 and 4 are connected and 5 and 6 are connected (polarity doesn’t matter).

Standard RJ 45 PIN assignment



Pin 1 – not used
Pin 2 – not used
Pin 3 – Transmit +
Pin 4 – Receive +
Pin 5 – Receive –
Pin 6 – Transmit –
Pin 7 – not used
Pin 8 – not used

ISDN Loopback plug



Pin 1 – not used
Pin 2 – not used
Pin 3 – Transmit +
Pin 4 – Receive +
Pin 5 – Receive –
Pin 6 – Transmit –
Pin 7 – not used
Pin 8 – not used

This test is a very simple test and should be done immediately after the configuration has been loaded to the KCS system and all the interfaces have been initialized correctly. TCOSS must be started, no cable is plugged into the TC34.

In this scenario, no cable attached to the TC34, the red LED should be on, the green LED should be off. Now connect the loopback plug to the TC34 RJ45 connection and the green LED should be ON immediately while the red LED must be off.

7.7 E1 Loopback Scenario with 2 TC34 Interfaces

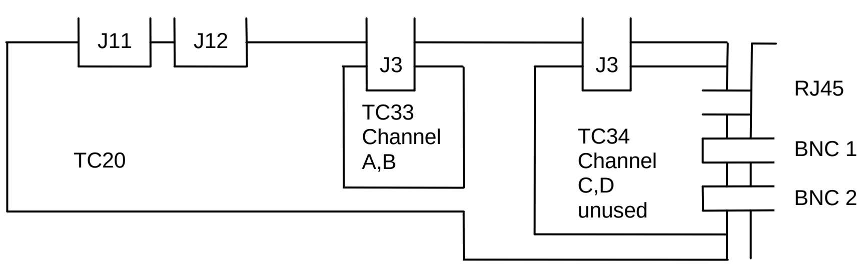
For testing purposes, hardware checks, configuration checks or simply to get a feeling about the primary rate connection procedure, two independent primary rate lines can be inter-connected. You would need the following hardware for such a scenario:

- Any KCS Model 2xx hardware, 1xTP80, 2xTC34, 2xTC33, 2xTC20, 2xTC34 PCM Bus cable (flat cable with 16 connectors), 2xTC20 link connection cables (10 way flat cable).

After you have all the necessary parts, connect the TC20, TC33 and TC34 together. This unit forms the first primary rate connection. Do the same with the second TC20, TC33 and TC34 interface to get the second primary rate connection.

Please note that you need special hardware revisions, jumper settings which are not described in detail here as they are described within the hardware documentation of the relevant parts.

After you have connected these parts you should have two units as displayed within the drawing below:



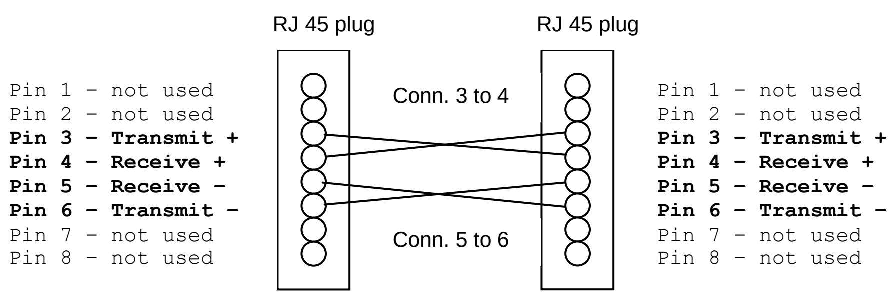
Now plug the TP80 into the first free ISA slot of the KCS model 2xx and be sure that address 150 HEX has been set. You can use any address, but with 150 HEX, the TCOSS slave number is defined as 1.1!

- Attach both primary rate units (TC20, TC33 and TC34) into the Model 2xx.
- Take the first TC20 link connection cable and connect TP80 – plug ST8, with first TC20 – plug J11.

- Take the second TC20 link connection cable and connect first TC20 – plug J12 with second TC20 – plug J11.
- Take the first PCM Bus cable (flat cable with 16 connectors) and connect one end to plug J3 on the TC33 located on the first TC20, connect the other end to the plug J3 on the TC34 located on the first TC20.
- Take the second PCM Bus cable (flat cable with 16 connectors) and connect one end to plug J3 on the TC33 located on the second TC20, connect the other end to the plug J3 on the TC34 located on the second TC20.

Now the hardware setup is ready, the only thing which is needed now is a crossed cable with two RJ45 plugs connected to each end – it should have the following setup:

RJ 45 E1 crossed cable connection



After the cable is ready, connect one end to the RJ45 connector located on the first TC34 and connect the other end on the second TC34's RJ45 connector.

Now the hardware setup is ready, the only thing missing is the software configuration which allows sending from one E1 to the other E1 connection and vice versa.

Therefore start the wconfig program, configure two ISDN channels located on the first TC20 as 1.1.L0 channels, configure two ISDN channels located on the second TC20 as 1.1.L1 channels, you should have following hardware/software assignment:

Channel	SW-Modul	Speed	HW-Modul	Slot	Slave
04)	IF		TC20-A	L0	1.1
05)	IF		TC20-B	L0	1.1
06)					
07)					
08)	IF		TC20-A	L1	1.1
09)	IF		TC20-B	L1	1.1

The channel allocation is free to choose, above channels 04 and 05 are used for the first primary rate connection while channel 08 and 09 are used for the second primary rate connection.

The configuration for primary rate connections is always performed via the so called “reference channel” (see KCS ISDN’s Reference Channel) which is the first channel of each primary rate connection. In the above configuration, channel 04 is the reference channel of the first primary rate connection while channel 08 is the reference channel of the second primary rate connection.

The first line which must be changed is config line 250, position 04 which defines whether the configuration is a basic rate or a primary rate connection.

One primary rate connection must simulate the “network side” while the other primary rate connection simulates the “user side”.

Network Side	**User side:**
:00 00 00 01 00 ,250	:00 00 00 01 00 ,250
:00 FF FF FF FF FF ,251	:00 FF FF FF FF FF ,251
'2, 252	'2, 252
:02 01 **01 00** 00 00 00 00 ,286	:02 01 **02 01** 00 00 00 00 ,286
:01 07 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 ,289	:**00** 07 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 ,289
:01 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00 00 ,291	:**00** 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00 00 ,291

The previous table shows the main differences between “network side” and “user side”. Only three config lines are different between both modes – nevertheless, they must be changed as described as otherwise the primary rate connection between both interfaces would not work. Changing these config lines can be done with the line editor only.

Do not configure both sides as “user side”! The green LED is on, but send attempts will lead immediately to error ISDN info 200 – no connection to local exchange”.

Nevertheless, if you have setup the KCS system as previously described, you can use the system to send FAX messages from primary rate connection 1 to 2 and vice versa. This allows to test most of the features offered by TCOSS without using a “real” primary rate connection.

8. KCS ISDN Configuration Lines

The following config lines below are the most important config lines for ISDN connections. They are described as these settings must be done via a line editor only which is not a comfortable way to do it but no other method exists with the current implementation.

The description does not contain any TAM (TCOSS Application Module) related config positions, config line 1 up to 50. It shows the most important TUM (TCOSS User module) specific config lines which are used frequently to establish an ISDN connection to customer's equipment.

8.1 ISDN Config Parameters

Config line 235 'FXI$ FXI$ FXC$,

Usage: 3 positions, prefix settings for inbound routing and user ID commands

Position 1…4 inbound prefix, unused positions filled with blanks

‘ empty, do not use inbound distribution
‘.FX inbound distribution via NN99
‘FXI$ inbound distribution via rr99, number normalization

Position 5 reserved, leave empty

Position 6…9 prefix for FAX commands, used with 8xxxx commands

‘+ get user ID and password from uu99 file
‘FXI$ get user ID from TCFW user profile

Position 10 reserved, leave empty

Position 11…14 prefix for originator in case caller ID is available

‘ empty, do always use TSI as originator (config line 42)
‘FXC$ If available, use caller ID as originator

Things to take care of:

- FXI has to be defined as service within KCS, inbound distribution is done via rr99 and inactive user addresses
- FXC has to be defined as service within KCS with a valid FAX service prefix. Default config is FAX$, for better understanding a different service name has been chosen!
- Examples of different usages described later!

Config line 236 :10 ,

Usage: 1 position, definition of maximum inbound number length

Position 1 Length of inbound DID or DDI number information. In this example, which is the default configuration for newly created channels, 16 digits are used for inbound distribution. Numbers which are longer than 16 digits are either cut off or treated as FAX command!

Config line 237 :01 01 01 ,

Usage: 3 positions, router function settings for DID or DTMF

Position 1 Password check enabled or disabled

00 check is disabled
01 check is enabled

Position 2 TSI, transmitted subscriber identification, check

00 TSI match is not required
01 TSI must match

Position 3 receiver number check

00 receiver number check within user profile disabled
01 receiver number check enabled

Config line 238, 239 and 240

Have the same meaning as config line 235, 236 and 237 – but are used for DTMF numbers and not for DID or DDI numbers.

Config line 249 ‘ ,

Usage: 1 position, definition of the ISDN reference channel for primary rate installations

Position 1 Definition of the reference channel, for primary rate connections, this is the channel where all other ISDN channels use the configuration from!

‘ , no reference channel has been defined
’04 wconfig channel 04 is used as reference channel

Please note that the reference channel itself can have it’s own channel number defined in this config line (pointing to itself) or this config position can be left empty for the reference channel – Both settings are possible and allowed.

All other primary rate channels which belong to the reference channel must contain the channel number of the reference channel as otherwise ISDN configurations are different!

Config line 250 :01 00 FF 00 00,

Usage: 5 positions, definition of connection types

Position 1 Connection type

00 point to point connection
01 point to multipoint connection

Position 2 Layer 3 protocol definition

00 EURO ISDN or Q.931 protocol
01 1TR6 protocol, old German standard
02 AT&T protocol for 4ESS switch type
03 QSIG protocol (since TCSP 7.47.07)

Position 3 TEI assignment procedure

00 setting for point to point connections
01…3F for non-automatic TEI value
FF automatic TEI assignment

Position 4 ISDN line type definition

00 Basic rate ISDN connection
01 Primary rate E1 ISDN connection
02 Primary rate T1 ISDN connection

Position 5 Coding standard

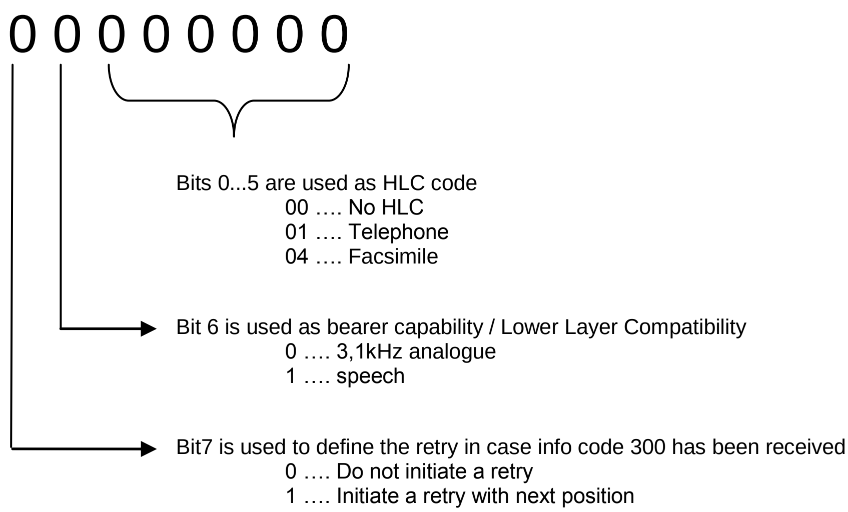
00 A-law coding, used in Europe
01 µ-law coding, used in USA and Japan

Config line 251 :00 FF FF FF FF FF ,

Usage: 6 positions, ISDN service definition

Position 1…6 **ISDN services** used for **outgoing FAX calls**, unused positions must be set to FF.

Each of these 6 positions above represents an 8 bit value, see the drawing below:



Up to now, the following combinations are supported within that config line:

00 01 04 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, do not use LLC frames, do not make any retries if receiver side aborts the call with ISDN info code 300

40 41 44 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, use additionally “speech” bearer capability within LLC frames, do not make retries if receiver side aborts the call with ISDN info code 300

80 81 84 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, do not use “speech” bearer capability within LLC frames, make retries if receiver side aborts the call with ISDN info code 300

C0 C1 C4 1) do not use HLC frames, 2) HLC=Telephone, 3) HLC=FAX G3, use additionally “speech” bearer capability within LLC frames, make retries if receiver side aborts the call with ISDN info code 300

Config line 252 ‘0,

Usage: 1 position, ISDN mode definition

Position 1 Definition of DDI or MSN

0 no DDI or MSN information available
1 MSN – multiple subscriber number, available
2 DDI – direct dialing in, available

Config line 253 ‘0,

Usage: 1 position, currency unit scaling

Position 1 currency unit scaling

0 report value in 1/1000 currency units
1 report value in 1/100 currency units
2 report value in 1/10 currency units
3 report value in currency units
4 report value in 10 currency units
5 report value in 100 currency units
6 report value in 1000 currency units

Config line 254 ' ,

Usage: 40 characters per line, number conversion table to route incoming/outgoing numbers

Format: <number type><search string>=<replace string>

<number type>
“1” line entry is only valid for DID/DDI numbers
“2” line entry is only valid for DTMF numbers
“3” line entry is valid for both numbers, DID/DDI and DTMF
“4” line entry is used to replace the CSI, Called Subscriber Identification
“8” line entry is used for all outgoing numbers
“@” line entry is used to convert the received caller ID (may include I, TI or TN9 before it is used as originator
“A” applies to redirecting number (incoming calls)
“C” applies to calling party number (outgoing calls)
“L” applies to call rerouting number (loop detection with outgoing calls)

<search string>
Any search sequence possible, ? as a single character wildcard or ~ as complete character replacement wildcard can be used, combination of both ? and ~ is possible

<replace string>
May be any desired character sequence.
If the replacement string starts with a T, an additional DTMF prompt will be generated.
If the replace string contains a % character, this character is replaced by the FAX Calling Party number
If the replace string starts with a V, the message is treated as Voice message and the remaining number part is used as “called number” (only with Line Server possible)

Examples of number conversions

‘1~=~ Use the received DID/DDI number for inbound distribution
‘1~=% Use the calling party number for inbound distribution, in this case be sure to set config line 236 to a higher value as otherwise characters are lost
‘4~=+43-1-86353-8~ Use always the dialed FAX number as CSI
‘1899=T generate a DTMF prompt after DID/DDI number 899 has been received
`173???=8000073???*0 get Document ??? (any number) from the FIS folder with session reversal
‘@TI~=*~ remove the TI string from a received caller ID

‘@~=~ leave received caller ID unchanged
'134=V131 incoming number 34 is treated as Voice call for number 131

Extended FAX number conversion table

The extended FAX number conversion has been implemented specially for 2nd dialing stage with DTMF and carrier support (e.g. MCI) for FAX send orders.

The number conversion table, which is defined by configuration lines 254 – 283 has been enhanced to use up to 10 different named variables for replacement. With this enhancement it is possible to:

- skip variable parts of the number or
- use any variable part more than one time or
- Change the order of variable parts.

To use this feature the syntax “**[~n]**” must be used instead or “**~**” and “**[?n]**” instead of “**?**”. n represents any number between “0” and “9” and is treated as variable name. The variable parts are inserted in the generated string at matching variable names as shown in the example below.

Conversion rule (x=type)	Before conversion	After conversion	Hint
x12[~1]555=12	12**3456**555	12555	Variable part has been removed
x12[~1]555=[~1]0[~1]	12**3456**555	**3456**0**3456**	Use variable part multiple time
x[~1]%00[~2]=[~2]#[~1]	**1234**%00**5678**	**5678**#**1234**	Change the order of variable parts

The old functionality with used just “~” or “?” is still available. **The new feature is not available for TSxx interfaces.**

Config line 255 up to 283 ' ,

Usage: 40 characters per line, number conversion table to route incoming/outgoing numbers

These lines have the same meaning as config line 254. You have 30 config lines 254 up to 283 available for the number conversion process. Nevertheless, config lines 276 up to 283 have the following default setting:

'8*~=TI~	,	276	conversion for outgoing number
'80~=0~	,	277	conversion for outgoing number
'8I~=I~	,	278	conversion for outgoing number
'8~=~	,	279	conversion for outgoing number
'@TI~=*~	,	280	remove the TI (international) from the caller ID
'@TN~=0~	,	281	remove the TN (national) from the caller ID
'@I~=I~	,	282	conversion for caller ID
'@~=~	,	283	conversion for caller ID

Config line 284 : 3C

Usage: 1 position, Call request timeout in seconds

00 ..FF 0 ... 255 seconds waiting time for identification of called station

Config line 285 : 05 10

Usage: 2 positions, DDI length and timeout definition

Position 1 DDI time-out in seconds

00 .. FF 0 seconds .. 255 seconds timeout

Position 2 Maximum count of DDI digits, only the configured count of digits is awaited

00 .. FF 0 digits up to 255 digits awaited

Config line 286 : 01 00 00 00 00 00 00 00 ,

Usage: 8 positions, ISDN specific protocol deviations

Position 1 ISDN call reference length

01 1 byte call reference length (Basic rate ISDN)
02 2 byte call reference length (Primary rate ISDN)

The main purpose is to have a “reference” value between calling party and called party.

Position 2 ISDN channel ID format

00 short format – used for basic rate interface
01 long format – used for primary rate interface
02 long channel identifier information element format (using slot map coding) to be used for Japanese INS-Net 1500 (PRI) if 01 does not work

Note that QSIG basic rate ISDN configurations need the long format too

Position 3 Outgoing B-channel negotiation method

00 any channel used, dedicated only for basic rate interface except for QSIG protocol
01 exclusive, for primary rate interface
02 preferred, for primary rate interface

Position 4 Call collision handling

00 A-side, TC34 selects B-channels for outgoing calls from lower end B1..
01 B-side, TC34 selects B-channels for outgoing calls from the upper end B30(B23)….

Position 5 Sending complete indicator in Outgoing Setup calls (Overlap sending) and LineServer incoming FAX/VOICE load balancing option (0th bit is the LSB bit, 7th the MSB bit)

Bit 0: 0 Send sending complete indicator with an outgoing SETUP (enbloc sending)
Bit 0: 1 Suppress sending of sending complete indicator with an outgoing SETUP (overlap sending).
Sending complete indicator must be suppressed in Japan and in the USA as not supported by ISDN networks there

Bit 1: 0 Suppress sending of LLC – lower layer compatibility with an outgoing SETUP
Bit 1: 1 Send LLC – lower layer compatibility with an outgoing SETUP

Bit 2: x reserved

Bit 3: 0 LS1 load balancing enabled (distribute incoming calls over DSP modules in a circular manner)
Bit 3: 1 LS1 load balancing disabled (incoming calls are routed to the free UIF channel with the lowest TCOSS channel number)

Position 6 Numbering Parameters for outgoing calls

Bit 0: 0 ISDN/Telephony numbering plan according to ITU recommendations E.164/E.163)
Bit 0: 1 Unknown numbering plan (should be used in Japan)

Bit 1-3 reserved
Bits 6-5 Screening indicator for the calling party number:
00 – user provided, not screened (default)
01 – user-provided, verified and passed
10 – user provided, verified and failed
11 – network provided
Bits 7-6 Presentation indicator for the calling party number:
00 – presentation allowed (default)
01 – presentation restricted
10 – number not available due to interworking
11 – reserved

Example: Value 10 hexadecimal denotes:
presentation allowed (00)
user-provided, verified and passed
ISDN/Telephony numbering plan

Position 7 AT&T service identification for outgoing calls

00 no specific service used for the call on the "cal-by-call" basis, must be provisioned for the whole T1 trunk
01 SDN service (software defined)
02 reserved
03 MEGACOM service
04 reserved
05 reserved
06 reserved
07 Long distance service

Position 8 Redirecting number for Fax calls and special setting for QSIG protocol (0^{th} bit is the LSB bit, 7^{th} the MSB bit)

Bit 0: 0 Redirecting number for fax calls (based upon CFU, CFB, CFNR) disabled
Bit 0: 1 Redirecting number for fax calls enabled. If available with an incoming fax call it is used for inbound routing instead of the DDI information, if not available then DDI is used instead.

Bit 1: 0 QSIG protocol variant „ISO/ECMA-QSIG" (default)
Bit 1: 1 QSIG protocol variant „ETSI-QSIG"
(this bit is relevant only for QSIG protocol configured in the line 250, position 2, ignored for any other protocols)
Supported since TCSP 7.50.01

Bit 2: 0 reserved

Bit 3: 0 QSIG B-channel numbering scheme using logical channel number (default).
Bit 3: 1 QSIG B-channel numbering scheme using E1 timeslot number (default).
(this bit is relevant only for QSIG protocol and E1 configurations, ignored for any other configuration type)
Supported since TCSP 7.50.01

Config line 287 : 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,

Usage: 16 positions, Layer 3 specific ISDN settings

Position 1…4 Allocation of B-channels, Bitwise map of B-channels is used. For details please refer to the fractional ISDN E1/T1 description

00 00 00 00 default value has the same meaning as 7F FF 7F FF for E1 installations
00 00 00 00 default value has the same meaning as 7F FF FF 00 for T1 installations

Position 5 timer value T302 (time defined in second units)

00 or 0F sets the default value for Euro ISDN and 1TR6. Please note that the value 00 is internally treated as 04. This timer is not used with AT&T protocol.

Position 6 timer value T303 (time defined in second units)

00 or 04 sets the default value for Euro ISDN and 1TR6
04 setting for AT&T protocol

Position 7 timer value T305 (time defined in second units)

00 or 1E sets the default value for Euro ISDN and 1TR6
04 setting for AT&T protocol

Position 8 timer value T308 (time defined in second units)

00 or 04 sets the default value for Euro ISDN and 1TR6
04 setting for AT&T protocol

Position 9 timer value T313 (time defined in second units)

00 or 04 sets the default value for Euro ISDN and 1TR6
04 setting for AT&T protocol

Position 10 timer value T317 (time defined in second units)

00 or 0A sets the default value for Euro ISDN and 1TR6
01 setting for AT&T protocol

Position 11..14 not used

Position 15..16 PBX Model

00 00 default (all PBXs except for those listed below)
00 01 Alcatel OmniPCX 4400 R6.x

Config line 288 ,

Usage: Comment line, currently not used, leave default setting

Config line 289 : 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00

Usage: 16 positions, Layer 2 specific ISDN settings

Position 1 Layer 2 side definition
00 KCS behaves as user side
01 KCS behaves as network side

Position 2 Window size k

00 or 01 Window size k=1, default setting for basic rate
07 Window size k=7, setting for primary rate

Position 3 Parameter N200 – retransmission counter

00 or 03 sets the default value for all protocols

Position 4 Parameter N202 – retransmission counter

00 or 03 sets the default value for all protocols

Position 5 Timer value T200 (time defined in 100 millisecond units)

00 or 0A sets the default value for all protocols

Position 6 Timer value T202 (time defined in 100 millisecond units)

00 or 0A sets the default value for all protocols

Position 7 Timer value T203 (time defined in 100 millisecond units)

00 or 64 sets the default value for Euro ISDN and 1TR6
96 sets the default value for AT&T protocol

Position 8 .. 16 not used at the moment, leave default settings

Config line 290 ,

Usage: Comment line, currently not used, leave default setting

Config line 291 : 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00

Usage: 16 positions, Layer 1 specific ISDN settings

Position 1 Primary rate Clock mode

00 KCS E1/T1 acts as slave mode
01 KCS E1/T1 acts as master mode

Position 2 Operation mode, Line coding and frame settings for E1 and/or T1 installations, Bit oriented value

HEX 00 in Binary writing: 0 0 0 0 0 0 0 0
Bit 8 …. Bit 1

Bit 1, value 1 00 …. E1 CRC-4 frame or T1 ESF - extended superframe
Bit 1, value 1 01 …. E1 FAS/NFAS double frame or T1 SF – superframe D4

Bit 2, value 2 00 …. T1 only, enable performance report message, PRM, according AT&T TR54016
Bit 2, value 2 01 …. T1 only, disable performance report message, PRM, according AT&T TR54016

Bit 3, value 4 00 …. T1 only, disable one-second PRM´s according ANSI T1.403
Bit 3, value 4 01 …. T1 only, enable one-second PRM´s according ANSI T1.403

Bit 4, value 8 00 …. T1 only, B8ZS line coding
Bit 4, value 8 01 …. T1 only, AMI with ZCS line coding

Used abbreviations:
CRC-4 E1 cyclical redundancy check 4 multiframe structure
B8ZS T1 Bipolar with 8 Zeros Substitution
AMI Alternate mark inversion
ZCS T1 Zero Code Suppression

Please note: You have to add the bitvalues according to the used coding and framing. The bitvalue for Bit 1 is 1, for Bit 2 is 2, for Bit 3 is 4 and for Bit 4 is 8.

So by using T1 Superframe with ZCS line coding and one second PRM's, the bitvalues for Bit 1, Bit 3 and Bit 4, 1 + 4 + 8 must be added. The resulting value, 13, must be defined as HEX value 0D within that position.

Position 3 .. 6 T1 transmit pulse shape in quarters of full wave, 3,6volts maximum

00 00 00 00 default value used for cable length of 0 .. 35 meters, internally treated as 14 14 14 14 HEX, output level is 3 volts peak

15 15 15 15 Output level of 3,150 Volts (an increase of 150 milli volts)

Position 7 **not used** at the moment, leave default settings

Position 8 **Reference clock priority** for **Line Server only (ignored on TC20 HW)**

00 Do never use this line as reference clock. This setting is recommended for Line Server Primary rate Loop tests (one TC24 interconnected with a crossed cable to simulate 2 PRI lines)
01 This line may be used as reference clock if the Layer 1 is active and configured as "clock slave"
02 same as value 01 but prefer this line as reference clock even if any other line

with priority 1 is currently used as reference clock. This is used for Line Server test environments with 1 PRI (TC24 loop connection) and 1 BRI (TC23). The BRI interface MUST have priority 2 in this scenario!

Position 9 **Supervision of BRI/PRI line status**
3C – the value of layer 1 supervision timer in seconds, default hex 3C (60 seconds).
00 - layer 1 supervision is deactivated (may only be necessary for some ISDN approvals in order not to confuse the ISDN testing equipment)

Position 10 **phase deviation select (PDS) - used with BRI only.**
Since TOCSS 7.63.01 the value of the PDS bit is both configurable and its default value has been changed to lower the phase jitter on the BRI line. (See error 11711)

00 (correspond with the new PDS value, which is used since TCOSS 7.63.01)

01 (use the same setting as before TCOSS 7.63.01)

Config line 292 ,

Usage: Comment line, currently not used, leave default setting

Config line 293 ,

Usage: Comment line, currently not used, leave default setting

Config line 294 ,

Usage: Comment line, currently not used, leave default setting

Config line 295 : 00 00 01 00 00 00 00 00

Usage: 8 positions, Various FAX/VOICE support options

Position 1 **Outgoing Voice support on that channel**
00 Outgoing Voice support disabled
01 Outgoing Voice support enabled

Position 2 **Use configured PABX prefix for outgoing voice calls**
00 Do not use the configured PABX prefix for outgoing voice calls
01 Use the configured PABX prefix for outgoing voice calls
Remark: The prefix refers to “automatic prefix for external sending” in fax config line 132.

Position 3 **Redirecting number for VOICE calls**
00 Redirecting number for incoming VOICE calls disabled
01 the Last Redirecting number for outgoing VOICE calls enabled

02 reserved
04 the Original Redirecting number for outgoing VOICE calls enabled

Note: The differentiation between Last and Original Redirecting number makes of course sense only during a multiple-call diversion. During a single call diversion there is only one redirecting number and this one is being used despite of having configured “Last” or Original” redirecting number.

Position 4 **Automatic FAX detection for incoming calls**
00 Automatic FAX detection disabled
other Automatic FAX detection enabled – e.g. “05” means that KCS tries to detect a CNG tone which would indicate an incoming FAX for 5 seconds

Position 5 **Call Transfer Mode**
00 (Internal) Call transfer via tromboning
01 (External) Call transfer by join to Alerting (preferred method).
02 (External) Call transfer by join to Connected (for compatibility reasons only)

Position 6 **Call Transfer via Call HOLD function**
00 Call Transfer without Call HOLD
01 Put the 1st Call on HOLD prior to the Call Transfer

Position 7..8 **not used**, leave default settings

Config line 296 : 00 00 00 00 00 00 00 00

Usage: 8 positions, status logging for incoming and outgoing VOICE calls

Position 1 **Type filter for logging entries** – This filter is a set of bits. If the corresponding bit is set, logging entries with that type are created

bit 0: filter for type = “Fax In”
bit 1: filter for type = “Fax Routing”
bit 2: filter for type = “Fax Scan”
bit 3: filter for type = “Fax Command”
bit 4: filter for type = “Fax Command (Session Reversal)”
bit 5: filter for type = “FIS Command”
bit 6: filter for type = “FIS Command (Session Reversal)”
bit 7: filter for type = “Voice In”

Position 2 **not used**, leave default setting

Position 3 **Result filter** – This filter is a set of bits. If the corresponding bit is set, logging entries with that type are created

bit 0: filter for result = “Error”
bit 1: filter for result = “No Fax”
bit 2: filter for result = “Ok”

bit 3: filter for result = “Ok (Tx)”
bit 4: filter for result = “Ok (Rx)”
bits 5-7 are reserved for future result values

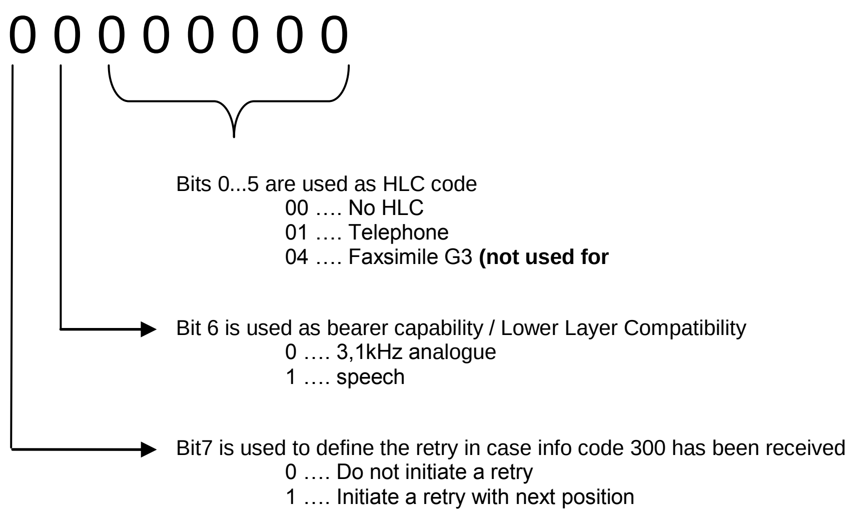
Position 4..8 **not used**, leave default setting

Config line 297 : 01 FF FF FF FF FF

Usage: 6 positions, ISDN HLC definition

Position 1…6 **ISDN services** used for **outgoing VOICE calls**, unused positions must be set to FF.

Each of these 6 positions above represents an 8 bit value, see the drawing below:



Up to now, the following combinations are supported within that config line:

00 01 1) do not use HLC frames, 2) HLC=Telephone, do not use LLC frames, do not make any retries if receiver side aborts the call with ISDN info code 300

40 41 1) do not use HLC frames, 2) HLC=Telephone, use additionally “speech” bearer capability within LLC frames, do not make retries if receiver side aborts the call with ISDN info code 300

80 81 1) do not use HLC frames, 2) HLC=Telephone, do not use “speech” bearer capability within LLC frames, make retries if receiver side aborts the call with ISDN info code 300

C0 C1 1) do not use HLC frames, 2) HLC=Telephone, use additionally “speech” bearer capability within LLC frames, make retries if receiver side aborts the call with ISDN info code 300

Config line 298 : 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00

Usage: 16 positions, reserved for incoming VOICE calls CODEC configuration, currently not used

Config line 299 : 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00

Usage: 16 positions, reserved for outgoing VOICE calls CODEC configuration, currently not used

8.2 ISDN Standard Configuration Examples

This chapter provides for examples of standard ISDN configurations for different protocols and access types. Please note that only protocol-specific settings are listed (no number conversions).

8.2.1 QSIG for BRI

Protocol variant: ISO/ECMA -QSIG (line 286, pos. 8 default value 00)
B-channel allocation: Preferred, B-side

Line	Configuration
250	`:00` **`03`** `00 00 00,  Pos.2 must be set to 03 for QSIG since TCSP 7.47.07`
252	`'2,  DDI for inbound routing`
286	`:02 01 02 01 00 00 00 00(*),`
287	`:00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,`
289	`:00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,`
291	`:00 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00,`

(*) – Protocol variant ETSI-QSIG may be setup by setting line 286, pos.8 to 02 since TCSP 7.50.01

8.2.2 QSIG for E1 PRI

Protocol variant: ISO/ECMA -QSIG (line 286, pos. 8 default value 00)
B-channel allocation: Preferred, B-side
E1 parameters: Clock-Slave, CRC4

Line	Configuration
250	`:00` **`03`** `00 01 00,  Pos.2 must be set to 03 for QSIG since TCSP 7.47.07`
252	`'2,  DDI for inbound routing`
286	`:02 01 02 01 00 00 00 00(*),`
287	`:00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,`
289	`:00 07 00 00 00 00 00 00 00 00 00 00 00 00 00 00,`
291	`:00 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00,`

(*) – Protocol variant ETSI-QSIG may be setup by setting line 286, pos.8 to 02 since TCSP 7.50.01

8.2.3 QSIG for T1 PRI

Protocol variant: ISO/ECMA-QSIG (line 286, pos. 8 default value 00)
B-channel allocation: Preferred, B-side
T1 parameters: Clock-Slave, ESF (extended superframe)

Line	Configuration
250	:00 **03** 00 02 01, Pos.2 must be set to 03 for QSIG since TCSP 7.47.07
252	'2, DDI for inbound routing
286	:02 01 02 01 00 00 00 00(*),
287	:00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,
289	:00 07 00 00 00 00 00 00 00 00 00 00 00 00 00 00,
291	:00 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00,

(*) – Protocol variant ETSI-QSIG may be setup by setting line 286, pos.8 to 02 since TCSP 7.50.01

8.2.4 DSS1 (EuroISDN) for E1 PRI

Protocol variant: Standard DSS1

B-channel allocation: Preferred, B-side

E1 parameters: Clock-Slave, CRC4

Line	Configuration
250	:00 00 00 01 00,
252	'2, DDI for inbound routing
286	:02 01 02 01 00 00 00 00,
287	:00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00,
289	:00 07 00 00 00 00 00 00 00 00 00 00 00 00 00 00,
291	:00 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00,

8.2.5 AT&T 4ESS for T1 PRI

Protocol variant: AT&T 4ESS custom

B-channel allocation: Preferred, B-side

T1 parameters: Clock-Slave, ESF (extended superframe)

Line	Configuration
250	:00 02 00 02 01,
252(*)	'1, "MSN" for inbound routing
254(*)	'3~=~,
286	:02 01 02 01 01 00 0**X** 00, 0**X**: configure AT&T service type!
287	:00 00 00 00 00 04 04 04 04 01 00 00 00 00 00 00,
289	:00 07 00 00 00 00 96 00 00 00 00 00 00 00 00 00,
291	:00 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00,

(*) In the USA the DDI configuration is not supported by the ISDN network. Therefore "MSN" must be configured in the line 252 in order to get the dialed number for inbound routing as a workaround for this restriction.
The only additional requirement is that all dialed numbers must be configured for incoming calls as "own" numbers somewhere in the lines 254-283, the simplest form that accepts all numbers is shown in the line 254 above. If this entry would be missing, all incoming calls would be ignored.

8.2.6 Japanese INS-Net 1500 PRI

Protocol variant: INS-Net 1500 based on DSS1 with some adaptations for Japan (lines 252, 286)
B-channel allocation: Preferred, B-side
T1 parameters: Clock-Slave, ESF (extended superframe)

Line	Configuration
250	`:00 00 00 02 01,`
252(*)	`'1, "MSN" for inbound routing`
254(*)	`'3~=~,`
286	`:02 01 02 01 01 01 00 00,`
287	`:00 00 00 00 00 04 04 04 04 01 00 00 00 00 00 00,`
289	`:00 07 00 00 00 00 96 00 00 00 00 00 00 00 00 00,`
291	`:00 00 00 00 00 00 00 01 00 00 00 00 00 00 00 00,`

(*) In Japan the DDI configuration is not supported by the ISDN network. Therefore "MSN" must be configured in the line 252 in order to get the dialed number for inbound routing as a workaround for this restriction.
The only additional requirement is that all dialed numbers must be configured for incoming calls as "own" numbers somewhere in the lines 254-283, the simplest form that accepts all numbers is shown in the line 254 above. If this entry would be missing, all incoming calls would be ignored.

8.3 ISDN Configuration Hints

8.3.1 QSIG Protocol Configuration

With TCSP releases older than 7.47.07 the QSIG protocol had to be setup as the standard DSS1 protocol (line 250, position 2 set to 00) with some modifications in the line 286.

<u>But since TCSP 7.47.07 QSIG protocol must be configured by the new QSIG protocol value (03) in the line 250, position 2</u>. For BRI access types it is supported by the Wconfig menus, but for the PRI it must be entered manually.

8.3.2 QSIG B-channel numbering

There is a difference in the B-channel numbering scheme between public network ISDN protocols (like DSS1) and private network QSIG protocols that are running via the PRI E1 trunk. While public ISDN protocols use the physical E1 trunk timeslot number in the B-channel negotiation procedures (timeslots 1-15, 17-31 for B channels 1-30), the QSIG lines use the logical B channel number during its negotiation (1-30).

On the other hand, there is no problem with B channel numbering using BRI access type or using PRI T1 trunk type.

Therefore if the QSIG protocol is configured (config line 250, 2nd position set to 03) the QSIG-conform B channel numbering scheme is being used by default.

<u>Note:</u> It is still possible in the real world, that some PBX systems may use the DSS1-like numbering scheme even though the QSIG protocol is in use. For example, Alcatel OmniPCX 4400 PBX with releases 3.2 and 4.2 has a configuration parameter "Logical channels" that may be set to 1__30 for QSIG, or 1__15 & 17__31 for DSS1-like B-channel numbering!).

In order to interconnect via QSIG protocol and E1 trunks also with those PBX systems that eventually do support only non QSIG-conform B-channel numbering scheme (using the physical E1 trunk timeslot number instead of logical B channel number) this channel numbering scheme can be enforced per configuration (config line 286, position 8)

Note: This configuration option is only a workaround for PBX systems using DSS1-like B channel numbering scheme for QSIG E1 PRI trunks. Please use it as the last possibility after it has been proved that the particular PBX can't be reconfigured to use the correct numbering scheme.

How to test whether the same numbering scheme is used on both sides:

1. Setup KCS to be the QSIG B-side (in order to force KCS to use the B30 channel for the first outgoing call)
2. Make one outgoing fax or voice call to any telephone
3. Answer the phone and check whether you hear the fax tones or voice data stream:
   if so, everything works fine
   If you hear silence or terrible noise, KCS and the PBX use different B channel numbering schemes. Verify whether the PBX is really setup for QSIG and whether there is any parameter defining the B channel numbering scheme to be used (like „Logical channels" parameter with Alcatel 4400).
   If it really looks like that the PBX uses wrong B channel numbering scheme, try to change it also on the KCS side using the bit 3 in the config line 286, position 8.

8.3.3 QSIG Protocol Variants ISO/ECMA-QSIG and ETSI-QSIG

QSIG protocol family is being standardized by 3 standardization bodies (ETSI, ECMA and ISO) and therefore PBX vendors usually support up to 3 different QSIG protocol variants for their PBX systems. They are most often being referred to as ETSI-QSIG, ECMA-QSIG and ISO-QSIG. While ECMA and ISO QSIG variants are normally compatible each with the other, the ETSI variant is compatible neither with ECMA nor ISO variant.

That is why also KCS's QSIG implementation supports 2 protocol variants ISO/ECMA-QSIG and ETSI-QSIG that are configurable with the config line 286, position 8.

The preferred QSIG protocol variant for KCS is the (default) ISO/ECMA one.

Please refer to the Model 305 PBX Integration manual for the information on which QSIG protocol variant is to be used for any particular PBX system that has already been validated by KCS.

If the PBX system has not been validated yet, ask the PBX technician to setup the PBX for ISO or ECMA QSIG variants and test the interconnection. Only if it does not work and the PBX supports also the ETSI-QSIG, setup ETSI QSIG variant on both sides and test again.

9. KCS ISDN Cable Connection Setup

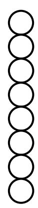
This is a very critical issue as our previous interface range TC20 with TC34 Primary rate ISDN has **another PIN connection** setup then the new interface range which is supported within the LineServer hardware TC24 Primary rate ISDN.

This has to be kept in mind whenever an existing customer with old style hardware (TSxx) is updated to new style hardware (TCxx). By doing so you **MUST REPLACE** the **EXISTING CABLE**!

9.1 TC20 Connection for TC33 and TC34 Interfaces

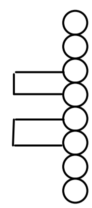
Up to now the following cable pinning has been used for basic rate / Primary rate ISDN connections:

Standard RJ 45 PIN assignment / TC20



Pin 1 – not used
Pin 2 – not used
Pin 3 – Transmit +
Pin 4 – Receive +
Pin 5 – Receive –
Pin 6 – Transmit –
Pin 7 – not used
Pin 8 – not used

ISDN TC34 Loopback plug

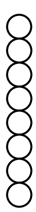


Pin 1 – not used
Pin 2 – not used
Pin 3 – Transmit +
Pin 4 – Receive +
Pin 5 – Receive –
Pin 6 – Transmit –
Pin 7 – not used
Pin 8 – not used

9.2 LineServer Connection for TC23 Basic Rate ISDN

This is identical with the old connection method as described above, nevertheless interconnection between two BRI lines is not possible, therefore no loopback cable can be used

Standard RJ 45 PIN assignment LS1 TC23

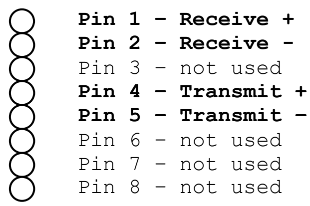


Pin 1 – not used
Pin 2 – not used
Pin 3 – Transmit +
Pin 4 – Receive +
Pin 5 – Receive –
Pin 6 – Transmit –
Pin 7 – not used
Pin 8 – not used

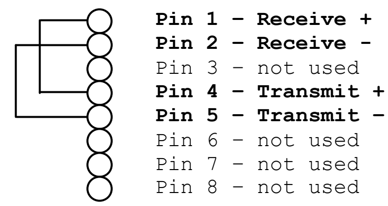
9.3 LineServer Connection for TC24 Primary Rate ISDN

This is definitely different to the old cable pinning. Therefore connection cable must be adapted/replaced whenever a customer is updated from old primary rate to new primary rate connection.

Standard RJ 45 PIN assignment / TC24



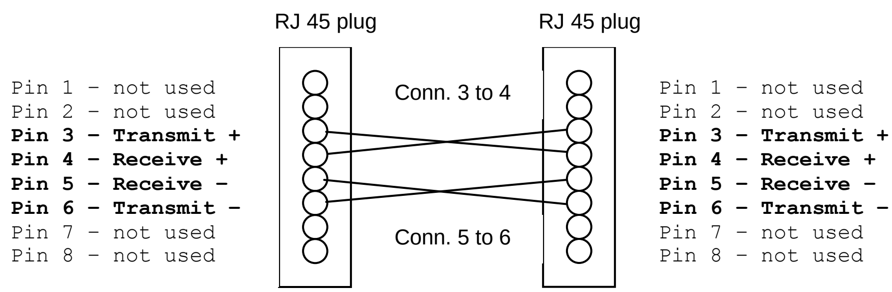
ISDN TC24 Loopback plug



9.4 Interconnecting two TC20/TC34 Primary Rate Connections

For testing purposes it might be useful to interconnect two TC20/TC34 Primary rate ISDN lines. This enables you to fully test sending/reception behavior without a “real” primary rate ISDN line (testing in office conditions).

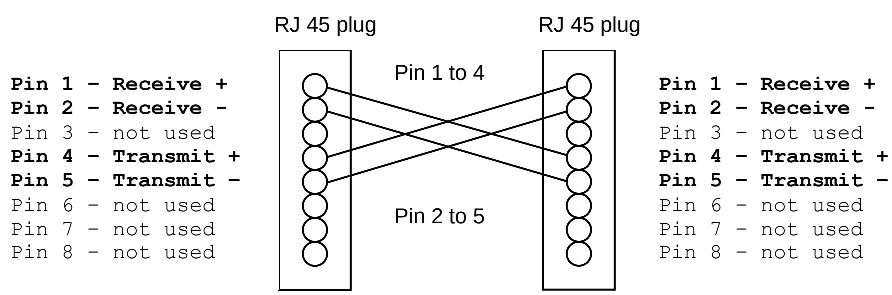
RJ 45 E1 crossed cable connection for TC20/TC34 ISDN



9.5 Interconnecting Two TC24 LineServer Primary Rate Lines

For testing purposes it might be useful to interconnect two TC24 Primary rate ISDN lines with a LineServer hardware. This enables you to fully test sending/reception behavior without a “real” primary rate ISDN line (testing in office conditions).

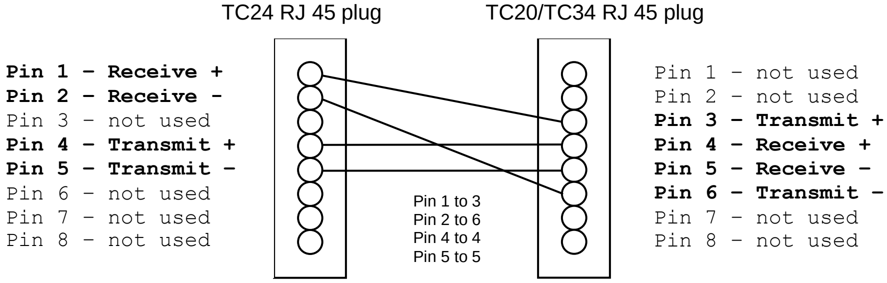
RJ 45 E1 crossed cable connection for TC24 ISDN



9.6 Interconnecting a TC24 PRI Interface with a TC20/TC34 PRI Interface

For testing purposes it might be useful to interconnect a TC24 Primary rate ISDN line with a TC20/TC34 Primary rate ISDN line. This enables you to fully test sending/reception behavior without a “real” primary rate ISDN line (testing in office conditions).

RJ 45 E1 crossed cable connection for TC24 to TC34 ISDN



10. KCS ISDN Trace Options

KCS offers various trace possibilities, like a plain ISDN trace, a FAX communication trace or a TAM/TUM trace which shows internal transfer between the user module and the application module. A combination of all three trace options can be used, depending on the problem you are facing at your customer.

10.1 Native ISDN Trace

When do you use such a trace: In case ISDN communication between KCS and the receiver cannot be established and the error code is an ISDN error code, usually with an ISDN info number attached, like ISDN info 34, ISDN info 200 …

To setup a native ISDN trace, you have to start your maketcoss configuration and add an additional channel.

- Start your customer configuration
- Select any free channel, double click on that channel and select a TR Tracer Module channel

```
15) TR Tracer Module
```

- Choose exactly the same slave number for the tracer module as you have used for the FAX module which causes the problem. **The TR tracer module for primary rate must use the slave and slot number where the TC34 belongs to**. Please note that a tracer module traces all channels which belong to a TC20 interface (the complete node), so 4 channels are traced with that module. In our example channel 04 is located on slot 1.1.L0, so the tracer module is also installed on 1.1.L0.

Channel	SW-Modul	Speed	HW-Modul	Slot	Slave
02)	TR		TC20	L0	1.1
03)					
04)	IF		TC20-A	L0	1.1

- Nothing more must be done, upload the configuration, and restart TCOSS. This must be done as a new channel has been installed.
- Now simply send a FAX to a number which causes problems.
- Trace output is written into the standard TCOSS trace file, located in c:\tcoss\traces

10.1.1 Native ISDN Trace Output

The native ISDN trace output looks like attached below

```
10:05:45.234 (390/428) N3/T1010 .001=          SETUP(4) >> NUM=999 CHIP=0
10:05:45.250 (390/428) N3/T988 .005=>> L=30 A=0201,C=0000 Setup(0001):  A1= 4=9090A3 18=A98381
70=81393939 7D=9184<
10:05:45.250 (390/428) N3/T1469 .013=<< L=4 A=0201,C=0102<
10:05:45.265 (390/428) N3/T1469 .009=<< L=14 A=0001,C=0002 Call-Proc(8001):  18=A98381<
10:05:45.281 (390/428) N3/T988 .002=>> L=4 A=0001,C=0102<
10:05:45.296 (390/428) N3/T1018 .002=                              << CALL PROC(4) CR = 1
10:05:45.312 (390/428) N3/T1010 .001=N_CALLPROC_IND(4) BCHAN=1
```

```
10:05:45.328 (390/428) N3/T1437 .001=arcofi_attach Chip=0,Timeslot=1
10:05:46.187 (390/428) N3/T1469 .939=<< L=9 A=0001,C=0202 Connect(8001): <
10:05:46.203 (390/428) N3/T988 .001=>> L=4 A=0001,C=0104<
10:05:46.218 (390/428) N3/T1018 .002=                                   << CONNECT(4)   CR = 1
10:05:46.234 (390/428) N3/T1010 .001=         CONN_ACK(4) >>
10:05:46.250 (390/428) N3/T1010 .000=N_CONN_CONF(4) BCHAN=1
10:05:46.265 (390/428) N3/T1438 .002=arcofi_attach Chip=0,Timeslot=1
10:05:46.281 (390/428) N3/T988 .003=>> L=9 A=0201,C=0204 Conn-Ack(0001): <
10:06:06.281 (390/428) N3/T1010 .001=         DISCONNECT(4) >> CA=16
10:06:06.296 (390/428) N3/T988 .003=>> L=13 A=0201,C=0404 Disconnect(0001):  8=8090<
10:06:06.312 (390/428) N3/T1469 .008=<< L=4 A=0201,C=0106<
10:06:06.328 (390/428) N3/T1469 .005=<< L=9 A=0001,C=0406 Release(8001): <
10:06:06.343 (390/428) N3/T988 .002=>> L=4 A=0001,C=0106<
10:06:06.359 (390/428) N3/T1018 .001=                                   << RELEASE(4) CR = 1 CA=-1
10:06:06.375 (390/428) N3/T1010 .002=  get_cause_aoc(4) CAUSE=-1 CURRENT AOC=FFFFFFFF GLOBAL AOC=0
10:06:06.390 (390/428) N3/T1010 .000=N_DISC_IND(4),CA=-1 AOC=0
10:06:06.406 (390/428) N3/T1010 .001=         REL COMPLETE(4) >>
```

The most important information which can be read from the trace is the ISDN setup procedure between KCS and the receiver.

Note: Whenever KCS **sends** ISDN commands, the sequence is **command >>**
Whenever KCS **receives** ISDN commands, the sequence in **<< command**

The most important information with a KCS outgoing setup message is the following string:

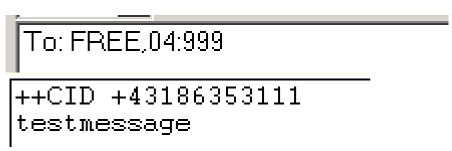
Setup(0001)**:** A1= 4=9090A3 **18**=A983**81** **70**=813**9**3**9**3**9** **7D**=91**84**<

This info element shows the **request of the B-channel**. In this case, B-channel 01 is requested. Please note that this is a bit value where Bit 8 is always 1 and does not count, Bit 1..7 show the channel allocation.

This info element shows the **dialed number as IA5 format** which is used within ISDN info elements as a standard.

This info element shows the **ISDN service** which is used from TOPCALL for this outgoing call. In this case 84 is used which defines a FAX G3 service

The same message is sent to the same destination number but now with the additional definition of a "calling party number" string. The message itself from KCS will look like



The KCS outgoing setup message looks like:

Setup(0002)**:** **A1**= 4=9090A3 **18**=A98381 **6C**=803**4**3**3**3**1**3**8**3**6**3**3**3**5**3**3**3**1**3**1**3**1** **70**=81393939 **7D**=9184

This info element shows the **sending complete** indicator

This info element shows the **request of the B-channel**

This info element shows **the "Calling Party Number"** in IA5 format

This info element shows the **"Called Party Number" - 999**.

This info element shows the **ISDN service**

The most important information with an incoming ISDN setup message is the following string:

```
Setup(28):  4=9090A3 18=89 1E=8083 +6C=118134333232323836333533363031 70=8131<
```

This info element shows **the "Calling Party Number"** in IA5 format

This info element shows the **"Called Party Number" - 1**. Please note that no sending complete indicator has been received – therefore other digits will follow

```
09:04:20.920 (21c/1c5) .065=72a:IFX DDI Number=1 OrigNum=TI4322286353601
09:04:21.250 (21c/1c5) .337=781:<< L=12 A=0200,C=1294 Info(28):  70=8137<


09:04:21.290 (21c/1c5) .002=72a:IFX DDI Number=17 OrigNum=TI4322286353601
09:04:21.671 (21c/1c5) .408=781:<< L=12 A=0200,C=1494 Info(28):  70=8131<

09:04:21.711 (21c/1c5) .011=72a:IFX DDI Number=171 OrigNum=TI4322286353601
09:04:21.721 (21c/1c5) .001=493:          CONNECT(18) >> CHIP=0

09:04:23.774 (21c/1c5) 2.00=72a:GETNUM (type 1) 171 => 171
09:04:23.774 (21c/1c5) .001=72a:CallerId: Converted TI4322286353601=>*4322286353601
```

From the trace above, the so called **overlapped reception mode** is active as we receive digit per digit. After we have received the complete sequence, 171 in this case, the CONNECT info is sent from KCS. As last step, both numbers, the DDI information 171 and the calling party number are converted based upon the "KCS number conversion table".

The purpose of the "call reference value" (config line 286 position 1) is to know exactly the relationship between ISDN commands. This is specially required if several ISDN channels send and receive at the same time. Basic rate ISDN uses a 1 byte call reference length, while primary rate uses a 2 byte call reference length.

As a rule the calling party uses a call reference value with the MSB (most significant bit) set to 0, while the called party uses as response a call reference value with the MSB set to 1.
In the example below, KCS receives an incoming message on a basic rate ISDN channel.

```
09:52:31.112 (48/137) 2.55=781:<< L=40 A=0200,C=86E6 Setup(0A):  4=9090A3 18=89 1E=8083
+ 6C=1181343934303533323839303438 70=8131<
09:52:31.132 (48/137) .002=483:>> L=4 A=0201,C=0188<
09:52:31.142 (48/137) .002=49b:       << SETUP(18)     CR=A L3ST=1

09:52:31.162 (48/137) .001=493:          SETUP_ACK(18) >>
09:52:31.172 (48/137) .004=483:>> L=11 A=0001,C=E688 Setup-Ack(8A):  18=89<
```

```
09:52:31.422 (48/137) .002=49b:       << INFO(18) CR = A
09:52:31.733 (48/137) .318=781:<< L=12 A=0200,C=8AE8 Info(0A):  70=8132<
09:52:31.743 (48/137) .002=49b:       << INFO(18) CR = A
09:52:32.053 (48/137) .316=72a:IFX DDI Number=112 OrigNum=TI494053289048
09:52:32.053 (48/137) .001=493:          CONNECT(18) >> CHIP=0
09:52:32.063 (48/137) .003=483:>> L=8 A=0001,C=E88C Connect(8A): <
09:52:32.073 (48/137) .021=781:<< L=4 A=0000,C=01EA<
09:52:32.133 (48/137) .052=781:<< L=8 A=0200,C=8CEA Conn-Ack(0A): <
```

The mandatory call reference value has been marked bold. KCS detects the setup message with call reference value 0A. It responses to this value with reference value 8A (MSB set to 1). Therefore 0A/8A build a pair of valid call reference values in the example above.

In the following example KCS sends a message via a primary rate connection.

```
15:35:06.836 (db/10d) 6.48=          SETUP(11) >> NUM=0021117462250 CHIP=0
15:35:06.886 (db/10d) .007=>> L=40 A=0001,C=0806 Setup(0002):  A1= 4=9090A3 18=A1839F +
70=8130303231313137343632323530 7D=9181<
15:35:06.926 (db/10d) .029=<< L=14 A=0201,C=060A Call-Proc(8002):  18=A9839F<
15:35:06.926 (db/10d) .001=>> L=4 A=0201,C=0108<
15:35:06.926 (db/10d) .004=       << CALL PROC(11) CR = 2
15:35:10.822 (db/10d) 3.88=<< L=21 A=0201,C=080A Connect(8002):
4C=21833231313137343632<
15:35:10.832 (db/10d) .002=>> L=4 A=0201,C=010A<
15:35:10.832 (db/10d) .003=       << CONNECT(11)   CR = 2
15:35:10.832 (db/10d) .004=          CONN_ACK(11) >>
15:35:10.842 (db/10d) .000=N_CONN_CONF(11) BCHAN=31
15:35:10.842 (db/10d) .003=arcofi_attach Chip=0,Timeslot=31
15:35:10.842 (db/10d) .004=>> L=9 A=0001,C=0A0A Conn-Ack(0002): <
```

The mandatory call reference value has been marked bold. KCS sends the setup message with call reference value 0002. The called party responses to this value with reference value 8002 (MSB set to 1). Therefore 0002/8002 build a pair of valid “call reference” values in the example above.

The next important information which can be “read” via an ISDN trace is the physical KCS channel which sends/receives the message which is displayed.

```
15:35:06.836 (db/10d) 6.48=          SETUP(11) >> NUM=0021117462250 CHIP=0
15:35:06.886 (db/10d) .007=>> L=40 A=0001,C=0806 Setup(0002):  A1= 4=9090A3 18=A1839F +
70=8130303231313137343632323530 7D=9181<
15:35:06.926 (db/10d) .029=<< L=14 A=0201,C=060A Call-Proc(8002):  18=A9839F<
15:35:06.926 (db/10d) .001=>> L=4 A=0201,C=0108<
15:35:06.926 (db/10d) .004=       << CALL PROC(11) CR = 2
15:35:10.822 (db/10d) 3.88=<< L=21 A=0201,C=080A Connect(8002):
4C=21833231313137343632<
15:35:10.832 (db/10d) .002=>> L=4 A=0201,C=010A<
15:35:10.832 (db/10d) .003=       << CONNECT(11)   CR = 2
15:35:10.832 (db/10d) .004=          CONN_ACK(11) >>
15:35:10.842 (db/10d) .000=N_CONN_CONF(11) BCHAN=31
15:35:10.842 (db/10d) .003=arcofi_attach Chip=0,Timeslot=31
15:35:10.842 (db/10d) .004=>> L=9 A=0001,C=0A0A Conn-Ack(0002): <
15:35:10.972 (db/10d) .103=<< L=4 A=0001,C=010C<
15:35:20.936 (db/10d) 10.0=>> L=4 A=0001,C=010B<
15:35:20.956 (db/10d) .009=<< L=4 A=0001,C=010D<
15:35:29.739 (db/10d) 8.78=          DISCONNECT(11) >> CA=16
15:35:29.739 (db/10d) .006=>> L=13 A=0001,C=0C0A Disconnect(0002):  8=8090<
15:35:29.769 (db/10d) .037=<< L=9 A=0201,C=0A0E Release(8002): <
```

```
15:35:29.779 (db/10d) .001=>> L=4 A=0201,C=010C<
15:35:29.779 (db/10d) .003=       << RELEASE(11) CR = 2 CA=-1
```

The ISDN commands displayed with capital letters show a value in brackets which describes the physical KCS channel defined via wconfig which sends/receives the message. It has nothing to do with the call reference value and is only KCS specific.

10.2 FAX Modem Trace

The FAX modem trace is generated by setting config lines 232 and 242 to the correct values.

Config line 232 traces the FXC part – so called FAX Converter part. This part in general is responsible for the complete coding of the FAX message like T4 or T6, it also generates the binary code which is then sent on the line. Furthermore it handles the FAX overlays, fonts, headerline, back reception, page and line breaks.

As a recommended value, for tracing options, this line 232 should always be set to 02 02 02 02. It's default value is 00 00 00 00

These kind of FAX traces should be used if the connection with the other fax machine could be established, but fax transmission or reception failed.

Install the Trace Tup channel and change the following config lines in the ISDN configuration of the specific channel which causes the problem:

```
config line 232 :02 02 02 02,
config line 242 :00 00 00 03,
```

in case you are facing the following FAX problems

- Outgoing calls are aborted with a error codes: XM, XN, XO, XP, XQ, XS, XT, XG
- Incoming calls with error codes: XV, XW, XY, XZ

If you have trouble with **tone detection** (e.g. you get error code XL instead of XJ, XU or XF) you should change

```
config line 232 :02 02 02 02,
config line 242 :00 00 02 03,
```

If you have troubles with the reception of **DDI information strings**, you should change:

```
config line 232 :02 02 02 02,
config line 242 :00 01 01 03,
```

Config line 242 traces the signal state machine, the FAX modem and the DID state machine and can generate a HEX dump of the reception document, see details below:

:00 00 00 00 00 00 00 00 00 00 00 00 ,242

Position 8..12, not used reserved for future usage

Position 7, LS1 DSP thread performance counter trace

00 …. No trace
01 …. DSP dump thread enabled

Position 6, LS1 Secondary data pump trace (DTMF detection)

00 …. No trace
01 …. Messages to data pump (useless here)
02 …. Messages from data pump
04 …. Transmitted PCM data (useless here)
08 …. Received PCM data

Position 5, LS1 Primary data pump trace (FAX / Voice data)

00 …. No trace
01 …. Messages to data pump
02 …. Messages from data pump
04 …. Transmitted PCM data
08 …. Received PCM data

Position 4, FAX modem trace

00 …. No modem trace
01…. Small trace output
02 …. More trace output
03 …. Recommended for FAX communication problems
04 …. More trace output than 3
05 …. More trace output than 4
06 …. Full FAX modem trace

Position 3, Signal State machine trace

00 …. No signal state machine trace
01 …. Low traceoutput (recommended)
02 …. High traceoutput

Position 2, DID state machine trace

00 …. No DID state machine trace (recommended)
01 …. DID state machine trace
02 …. Run in FAX test mode (direct function calls)

Position 1, HEX dump reception

00 …. Normal reception (recommended)
01 …. Received docs are stored as HEX dump

10.2.1 Native FAX Modem Trace Output

The native FAX modem trace output looks like attached below


```
16:13:50.484 (390/428) N3/T991 .002=          SETUP(4) >> NUM=999 CHIP=0
16:13:50.500 (390/428) N3/T969 .006=>> L=30 A=0201,C=0000 Setup(0001):  A1= 4=9090A3 18=A98381
70=81393939 7D=9184<
16:13:50.515 (390/428) N3/T1469 .011=<< L=4 A=0201,C=0102<
16:13:50.531 (390/428) N3/T1469 .010=<< L=14 A=0001,C=0002 Call-Proc(8001):  18=A98381<
16:13:50.546 (390/428) N3/T969 .001=>> L=4 A=0001,C=0102<
16:13:50.562 (390/428) N3/T1000 .002=                                  << CALL PROC(4) CR = 1
16:13:50.578 (390/428) N3/T991 .001=N_CALLPROC_IND(4) BCHAN=1
16:13:50.593 (390/428) N3/T1437 .001=arcofi_attach Chip=0,Timeslot=1
16:13:51.359 (390/428) N3/T1469 .887=<< L=9 A=0001,C=0202 Connect(8001): <
16:13:51.375 (390/428) N3/T969 .001=>> L=4 A=0001,C=0104<
16:13:51.390 (390/428) N3/T1000 .002=                                  << CONNECT(4)   CR = 1
16:13:51.406 (390/428) N3/T991 .001=          CONN_ACK(4) >>
16:13:51.421 (390/428) N3/T991 .000=N_CONN_CONF(4) BCHAN=1
16:13:51.437 (390/428) N3/T1438 .002=arcofi_attach Chip=0,Timeslot=1
16:13:51.453 (390/428) N3/T969 .003=>> L=9 A=0201,C=0204 Conn-Ack(0001): <
16:13:51.468 (390/428) N3/T1378 .000=04:MOD RxGain1 1
16:13:51.484 (390/428) N3/T1378 .003=04:MOD sound 1100, 50
16:13:51.500 (390/428) N3/T1469 .005=<< L=4 A=0201,C=0104<
16:13:51.875 (390/428) N3/T1378 .501=signal start=Z,hist=0
16:13:51.890 (390/428) N3/T1378 .000=04:MOD tondet
16:13:51.921 (390/428) N3/T1378 .047=Sig at 1 Jump to Z
16:13:53.453 (390/428) N3/T1378 1.53=Sig at 154 Jump to z
16:13:56.468 (390/428) N3/T1378 3.00=Sig at 454 Jump to -
16:13:56.484 (390/428) N3/T1378 .001=04:MOD idle2
16:13:56.500 (390/428) N3/T1378 .000=04:MOD sound 1100, 50
16:13:56.968 (390/428) N3/T1378 .506=signal start=Z,hist=0
16:13:56.984 (390/428) N3/T1378 .000=04:MOD tondet
16:13:57.015 (390/428) N3/T1378 .047=Sig at 1 Jump to Z
16:13:57.109 (390/428) N3/T1378 .091=Sig at 9 Jump to =
16:13:57.125 (390/428) N3/T1378 .001=04:MOD idle2
16:13:57.140 (390/428) N3/T1378 .000=04:MOD rx_start1 1 0 0
16:13:57.156 (390/428) N3/T1378 .021=04:MOD rx_start2 4
16:13:57.171 (390/428) N3/T1378 .011=04:MOD AGC Gain = 30294
16:13:57.250 (390/428) N3/T1378 .107=04:MOD mod_rx_stat Activate Rx Attenuator (30296)
16:13:57.265 (390/428) N3/T1378 .001=04:MOD RxGain1 0
16:13:58.468 (390/428) N3/T1378 1.22=04:MOD-Rx ok(23) = rCSI Fax:              +43-1-08
16:13:58.734 (390/428) N3/T1378 .266=04:MOD-Rx ok(07) = rDIS (V.27/29/33/17,ECM) 00EEB844
16:13:58.812 (390/428) N3/T1378 .072=04:MOD idle2
16:13:58.828 (390/428) N3/T744 .002=ConIn>TAMIO 2:Opt=fine, fkz=0:0000013438,D=011121-
161349,Retr=0
16:13:58.859 (390/428) N3/T744 .043=ConRs<TAMIO 2:Obj=2,Err=0,scanmode=0
16:13:58.875 (390/428) N3/T746 .001=BAPAG<TAMIO 2:page=0,Format=A,Landscape=0,Auto=0
16:13:58.890 (390/428) N3/T742 .004=BAPAG>TAMIO 2:page=1,Format=A,Landscape=0,Auto=0
16:13:58.906 (390/428) N3/T1378 .018=04:MOD tx_start1 1 0 0
16:13:59.031 (390/428) N3/T746 .152=BOPAG<TAMIO 2:tm=78,next bm=78,bm=120,length=944,witdh=1615,
16:13:59.046 (390/428) N3/T743 .002=BBLK >TAMIO 2:Typ=TCI,Res=2,Pos=(0,0,0,1,0),
16:13:59.234 (390/428) N3/T746 .189=BBLK <TAMIO
2:Typ=Text,Res=1,Pos=(0,0,0,1,0),,Fnt=1,lm=55,lpi=16,cpi=17,cm
16:13:59.250 (390/428) N3/T746 .000=     ax=95,bsoff=5
16:13:59.343 (390/428) N3/T746 .117=EBLK <TAMIO 2:Typ=Text
16:13:59.359 (390/428) N3/T746 .001=EPgRq<TAMIO 2:last=1,fill=0,EndOfDoc=1
16:13:59.937 (390/428) N3/T1378 .592=04:MOD tx_start2
16:13:59.953 (390/428) N3/T1378 .002=04:MOD-Tx ok(23) = tTSI Fax:              +43-1-04
16:13:59.968 (390/428) N3/T1378 .001=04:MOD-Tx ok(07) = tDCS (V.17-144,ECM) 0022F844
16:14:00.968 (390/428) N3/T1378 1.04=04:MOD idle2
16:14:00.984 (390/428) N3/T1378 .000=04:MOD long_train
16:14:01.000 (390/428) N3/T1378 .002=04:MOD tx_start1 0 A 1
16:14:01.015 (390/428) N3/T1378 .019=04:MOD tx_start Wait=754
16:14:01.375 (390/428) N3/T969 .372=>> L=4 A=0201,C=0105<
16:14:01.390 (390/428) N3/T1469 .001=<< L=4 A=0001,C=0105<
16:14:01.406 (390/428) N3/T969 .002=>> L=4 A=0001,C=0105<
16:14:01.421 (390/428) N3/T1469 .005=<< L=4 A=0201,C=0105<
16:14:02.656 (390/428) N3/T1378 1.27=04:MOD tx_start2
16:14:04.203 (390/428) N3/T1378 1.55=04:MOD idle2
16:14:04.218 (390/428) N3/T1378 .001=04:MOD rx_start1 1 0 0
16:14:04.234 (390/428) N3/T1378 .022=04:MOD rx_start2 4
16:14:05.609 (390/428) N3/T1378 1.37=04:MOD-Rx ok(03) = rFTT
```

```
16:14:05.687 (390/428) N3/T1378 .082=04:MOD idle2
16:14:05.703 (390/428) N3/T1378 .000=04:MOD tx_start1 1 0 0
16:14:06.734 (390/428) N3/T1378 1.04=04:MOD tx_start2
16:14:06.750 (390/428) N3/T1378 .001=04:MOD-Tx ok(23) = tTSI Fax:              +43-1-04
16:14:06.765 (390/428) N3/T1378 .002=04:MOD-Tx ok(07) = tDCS (V.17-120,ECM) 002AF844
16:14:07.765 (390/428) N3/T1378 1.04=04:MOD idle2
16:14:07.781 (390/428) N3/T1378 .000=04:MOD long_train
16:14:07.796 (390/428) N3/T1378 .001=04:MOD tx_start1 0 9 1
16:14:07.812 (390/428) N3/T1378 .017=04:MOD tx_start Wait=812
16:14:09.437 (390/428) N3/T1378 1.65=04:MOD tx_start2
16:14:11.000 (390/428) N3/T1378 1.55=04:MOD idle2
16:14:11.015 (390/428) N3/T1378 .001=04:MOD rx_start1 1 0 0
16:14:11.031 (390/428) N3/T1378 .022=04:MOD rx_start2 4
16:14:11.390 (390/428) N3/T969 .360=>> L=4 A=0201,C=0105<
16:14:11.406 (390/428) N3/T1469 .001=<< L=4 A=0001,C=0105<
16:14:11.421 (390/428) N3/T969 .001=>> L=4 A=0001,C=0105<
16:14:11.437 (390/428) N3/T1469 .005=<< L=4 A=0201,C=0105<
16:14:12.390 (390/428) N3/T1378 1.00=04:MOD-Rx ok(03) = rCFR
16:14:12.484 (390/428) N3/T1378 .081=04:MOD idle2
16:14:12.562 (390/428) N3/T1378 .075=04:MOD tx_start1 1 9 0
16:14:13.171 (390/428) N3/T1378 .618=04:MOD tx_start2
16:14:13.796 (390/428) N3/T1378 .625=04:MOD idle2
16:14:13.812 (390/428) N3/T1378 .001=04:MOD tx_start1 1 0 1
16:14:13.828 (390/428) N3/T1378 .020=04:MOD tx_start Wait=774
16:14:13.843 (390/428) N3/T632 .000=CPU Usage (3): 27.43% (HighPri 5.68%, 521us)
16:14:14.890 (390/428) N3/T1378 1.07=04:MOD tx_start2
16:14:14.906 (390/428) N3/T1378 .001=04:MOD-Tx ok(07) = tPPS-tEOP 2F000002
16:14:15.234 (390/428) N3/T1378 .352=04:MOD idle2
16:14:15.250 (390/428) N3/T1378 .000=04:MOD rx_start1 1 0 0
16:14:15.265 (390/428) N3/T1378 .025=04:MOD rx_start2 4
16:14:16.718 (390/428) N3/T1378 1.45=04:MOD-Rx ok(03) = rMCF
16:14:16.796 (390/428) N3/T1378 .081=04:MOD idle2
16:14:16.828 (390/428) N3/T743 .024=EBLK >TAMIO 2:Typ=TCI,Text=230,
16:14:16.843 (390/428) N3/T745 .019=EPgIn>TAMIO 2:Obj=0,Err=0,fkz=,DSNR=021-001,Skip=0,AcknPg=1
16:14:16.921 (390/428) N3/T745 .077=EPgRs<TAMIO 2:Obj=2,Err=0,pg_qual=0
16:14:16.937 (390/428) N3/T746 .001=DisRq<TAMIO 2:Obj No.=2, Err No.=0
16:14:16.953 (390/428) N3/T1378 .000=04:MOD tx_start1 1 0 0
16:14:17.968 (390/428) N3/T1378 1.04=04:MOD tx_start2
16:14:17.984 (390/428) N3/T1378 .000=04:MOD-Tx ok(03) = tDCN
16:14:18.218 (390/428) N3/T1378 .247=04:MOD idle2
16:14:18.234 (390/428) N3/T991 .001=N_DISC_REQ(4)
16:14:18.250 (390/428) N3/T991 .000=          DISCONNECT(4) >> CA=16
16:14:18.265 (390/428) N3/T969 .003=>> L=13 A=0201,C=0404 Disconnect(0001):  8=8090<
16:14:18.281 (390/428) N3/T1469 .009=<< L=4 A=0201,C=0106<
16:14:18.296 (390/428) N3/T1469 .006=<< L=9 A=0001,C=0406 Release(8001): <
16:14:18.312 (390/428) N3/T969 .002=>> L=4 A=0001,C=0106<
16:14:18.328 (390/428) N3/T1000 .003=                                  << RELEASE(4) CR = 1 CA=-1
16:14:18.343 (390/428) N3/T991 .001=  get_cause_aoc(4) CAUSE=-1 CURRENT AOC=FFFFFFFF GLOBAL AOC=0
```

You see clearly several state machine entries, TAMIO specific settings and MODEM RX (receive) and MODEM TX (transmit) commands.

The general FAX sending procedure between "sender" and "receiver" is explained below:

Calling unit (Sender)	Called Unit (Receiver)	Short explanation
CNG		Calling tone: 1100 Hz, 500 msec on / 3 sec. Off indicate non-speech terminal
PHASE A	CED	Called station ID: 2100 Hz, 2,6sec < on < 4 sec
	DIS	Digital ID Signal: 300BPS FSK, HDLC format
DCS		Digital command signal: 300BPS FSK, HDLC format
TCF		Training check: High speed train followed by 1,5sec of zeros
PHASE B	CFR	Confirmation to receive: 300BPS FSK, HDLC format
MESSAGE **PHASE C**		Transmits Document
EOM		End of message: 300BPS FSK, HDLC format EOP, MPS or PRI-Q may be sent
PHASE D	MCF	Message confirmation: 300BPS FSK, HDLC format Post-message response of RTP, RTN, PIP or PIN may be sent

FSK Frequency shift keying (Modulation type)
HDLC High level data link control (standard procedure for data communication)

PHASE A: Is defined as "call establishment" between sender and receiver (CNG and CED)

PHASE B: Is defined as "pre-message procedure". It consists of the handshake. One machine sends an identification signal, the other machine responds with a command signal. Training check is done with V.21 and 300bps FSK modulation between sender and receiver

PHASE C: Occurs after both sender and receiver have set up for the high speed configuration that was decided upon in phase B. Message transfer occurs here

PHASE D: Is defined as "post message phase". It uses FSK and HDLC format.

10.2.2 Index of Used HDLC Abbreviations Used in Recommendation T.30

Abbreviation	Function	Signal format
CED	called station identification	2100 Hz
CFR	confirmation to receive	X010 0001 1850 or 1650 Hz for 3 sec.
CRP	command repeat	X101 1000
CIG	calling subscriber identification	1000 0010
CNG	calling tone	1100 Hz for 500 msec
CSI	called subscriber identification	0000 0010
CTC	continue to correct	X100 1000
CTR	response to continue to correct	X010 0011
DCN	disconnect	X101 1111
DCS	digital command signal	X100 0001
DIS	digital identification signal	0000 0001
DTC	digital transmit command	1000 0001
EOM	end of message	X111 0001 1100 Hz
EOP	end of procedure	X111 0100
EOR	end of retransmission	X111 0011
ERR	response for end of retransmission	X011 1000
FCD	facsimile coded date	0110 0000
FCF	facsimile control field	
FIF	facsimile information field	
FTT	failure to train	X010 0010
GC	group command	1300 Hz for 1,5 - 10 sec 2100 Hz for 1,5 - 10 sec
GI	group identification	1650 or 1850 Hz
HDLC	high level data link control	
LCS	line conditioning signals	1100 Hz
MCF	message confirmation	X011 0001 1650 or 1850 Hz
MPS	multi page signal	X111 0010
NSC	non standard facilities command	1000 0100
NSF	non standard facilities	0000 0100
NSS	non standard setup	X100 0100
PIN	procedural interrupt negative	X011 0100
PIP	procedural interrupt positive	X011 0101
PIS	procedure interrupt signal	462 Hz for 3 sec
PPS	partial page signal	X111 1101
PPR	partial page request	X011 1101
PRI-EOM	procedure interrupt-EOM	X111 1001
PRI-EOP	procedure interrupt-EOP	X111 1100
PRI-MPS	procedure interrupt-MPS	X111 1010
RCP	return to control for partial page	0110 0001
RNR	receive not ready	X011 0111
RR	receive ready	X111 0110
RTN	retrain negative	X011 0010
RTP	retrain positive	X011 0011
TCF	training check	zeros for 1,5 sec
TSI	transmitting subscriber identification	X100 0010

10.3 TAM/TUM Trace Output

It is possible to activate a TAM/TUM trace output which shows communication between the application module and the user module part (used for all channels working with the dot-dot Interface).

These are all channels except UC0 channels. The trace can be activated for each channel independently with config line 32 of the channel which should be traced. The trace output is written into the standard TCOSS trace file located in C:\tcoss\trace\tcossx.trc. It does not rely on any other trace setting.

‘0 ,32

Position 1, TAM/TUM tracer options

0 …. No trace activated
1 …. minimum trace (only commands/response no text lines)
2 …. standard trace (commands/responses, truncated text lines, recommended)
3 …. enhanced trace (as ‘2’ but do not truncate text lines and additional receiver ready trace)
4 …. full trace (includes empty commands and polls, may be very big!)
5 …. full hex trace

In most cases the standard trace is appropriate. The truncated lines are useful to reduce the trace data created by TCI image blocks. These blocks are usually using the maximum possible line length of 254 characters.

The TAM/TUM trace shows the reception of FAX, the inbound distribution (NN99, rr99, originator based routing ….), the received responses whether the reception user exists or not and the reception quality like “eyequality” or “reception speed”.

It should be used whenever you have problems with inbound distributions to a specific user ID.

10.3.1 Native TAM/TUM Trace Output

A native TAM/TUM trace output which shows the reception of a FAX message with inbound routing via rr99 and "standard trace setting" will look like attached below: Please note that for better understanding the ISDN trace has been enabled too.

Example 1 – Reception of FAX with rr99 inbound distribution

```
09:11:15.093 (390/428) N3/T1018 .002=          << SETUP(4)     CR = 1 L3ST=1
09:11:15.125 (390/428) N3/T1010 .001=         CALL_PROC(4) >>
09:11:16.265 (390/428) N3/T1302 .000=IFX DDI Number=999 OrigNum=
09:11:16.281 (390/428) N3/T1010 .001=         CONNECT(4) >> CHIP=0
09:11:16.359 (390/428) N3/T1018 .001=         << CONN ACK(4)  CR = 1
09:11:18.265 (390/428) N3/T1301 2.00=GETNUM (type 1) 999 => 999
09:11:18.265 (390/440) 04:TAM   ICmd 2//2CHECK,N=FXI$999<
09:11:18.281 (390/440) 04:TAM Resp 2105 TCTECH  : <
09:11:25.875 (390/440) 04:TAM   ICmd 2//1LOGON,TYP=1,AUTOR=+43-1-08                  <
09:11:25.906 (390/440) 04:TAM Resp 2101 ATF0018     011122 091125 OK<
09:11:25.937 (390/440) 04:TAM   ICmd 2//2S,N=FXI$999<
09:11:25.953 (390/440) 04:TAM Resp 2100 OK<
09:11:31.015 (390/440) 04:TAM   Txt 2++FX2<
09:11:31.015 (390/428) N3/T748 4.74=Decode Mode = 1 (TestSwitch=0)
09:11:31.437 (390/440) 04:TAM   Txt 2TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq+
09:11:31.578 (390/440) 04:TAM   Txt 27eyHsldkpsdwx3Zyq,kHcInvh74vV2UQ9IaqBIE+
09:11:31.734 (390/440) 04:TAM   Txt 2SSBKT1z2UPrIeyylU,kHZZV2BaVdXZDhZAVSpZD+
09:11:31.765 (390/440) 04:TAM   Txt 2JbUViVDrHSkNVI4jw8PQ(qWUDqEeugIPXPXQJY6+
09:11:31.796 (390/440) 04:TAM   Txt 2q,TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq,T+
09:11:33.328 (390/440) 04:TAM   Txt 2aMg,lW3(gdRv5RxXA7S53c7um3g,knHwXgpx5Rx+
09:11:33.359 (390/440) 04:TAM   Txt 2++TXT 110, 0 0 14400+ EQ=1 -9dBm<
09:11:33.390 (390/440) 04:TAM   ICmd 2//1ESEITE,DSNR=   -001<
09:11:33.421 (390/440) 04:TAM Resp 2100 OK<
09:11:36.265 (390/428) N3/T1018 .002=           << DISCONNECT(4) CR = 1 CA=16
09:11:36.265 (390/440) 04:TAM   ICmd 2//1LOGOFF,FC=  ,ENDE=0,AUTOR=+43-1-08<
09:11:36.281 (390/440) 04:TAM Resp 2100 OK<
09:11:36.281 (390/428) N3/T1010 .000=         RELEASE(4) >>
09:11:36.359 (390/428) N3/T1018 .002=         << REL COMPL(4) CR = 1 CA=-1
```

Example 2 – Reception of FAX with rr99 inbound distribution and caller ID (without ISDN trace)

```
09:29:20.578 (390/458) 08:TAM   ICmd 2//2CHECK,N=FXI$999<
09:29:20.593 (390/458) 08:TAM Resp 2105 TCTECH  : <
09:29:28.171 (390/458) 08:TAM   ICmd 2//1LOGON,TYP=1,AUTOR=+43-1-04             <
09:29:28.203 (390/458) 08:TAM Resp 2101 ATF0019     011122 092928 OK<
09:29:28.250 (390/458) 08:TAM   ICmd 2//2S,N=FXI$999,OR=FXC$43186353111<
09:29:28.265 (390/458) 08:TAM Resp 2100 OK<
09:29:33.312 (390/458) 08:TAM   Txt 2++FX2<
09:29:33.734 (390/458) 08:TAM   Txt 2TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq,TZq+
09:29:33.890 (390/458) 08:TAM   Txt 2Y7s5VA,kHcInvh74vV2Q9lICqBIEgRPZUg5euQO+
09:29:33.921 (390/458) 08:TAM   Txt 2Ush4eHq7KuyXrjiZA7AtKurshwsgvunR0iAt88m+
09:29:34.031 (390/458) 08:TAM   Txt 2wdY8HqFHC6j(98FyAhLEgS3McBXEJxwjiQJbmPB+
09:29:34.062 (390/458) 08:TAM   Txt 2TZq,TZq,TZq,TZq,TZq,kn1LTC0,lXBcmFo,lXB+
09:29:35.625 (390/458) 08:TAM   Txt 2M,bhU29A,blIjb1,blQ23U,TZq,TZq,TZq,TZq,+
09:29:35.656 (390/458) 08:TAM   Txt 2++TXT 110, 0 0 14400+ EQ=2 -9dBm<
09:29:35.687 (390/458) 08:TAM   ICmd 2//1ESEITE,DSNR=   -001<
09:29:35.718 (390/458) 08:TAM Resp 2100 OK<
09:29:38.531 (390/458) 08:TAM   ICmd 2//1LOGOFF,FC=  ,ENDE=0,AUTOR=+43-1-04     <
09:29:38.546 (390/458) 08:TAM Resp 2100 OK<
```

Example 3 – Reception of FAX with unsuccessful rr99 inbound distribution and caller ID

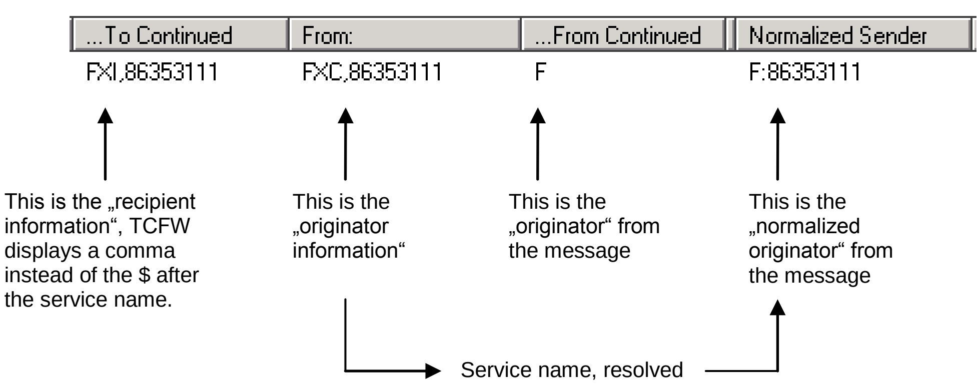
```
09:39:55.406 (390/428) N3/T1010 .001=         CONN_ACK(4) >>
09:39:57.390 (390/458) 08:TAM   ICmd 2//2CHECK,N=FXI$123<
09:39:57.406 (390/458) 08:TAM Resp 2405 bad number<
09:39:58.687 (390/428) N3/T1018 .001=         << DISCONNECT(4) CR = 3 CA=16
```

After the ISDN connection has been established successfully, a //2CHECK is sent to get the user ID for the DDI number 123. As no user exists and the default recipient has been disabled within the rr99 file, KCS does not accept this message.

Example 4 – Reception of FAX with originator based routing

```
10:13:48.906 (390/428) N3/T1469 4.56=<< L=37 A=0001,C=0000 Setup(0001):  A1=
............      4=9090A3 18=A1839F +6C=803836333533313131 70=81313233<
10:13:49.953 (390/428) N3/T1302 .001=IFX DDI Number=123 OrigNum=86353111
10:13:51.937 (390/440) 04:TAM   ICmd 2//2CHECK,N=FXI$86353111<
10:13:51.937 (390/440) 04:TAM Resp 2105 DUMB    : <
10:13:51.953 (390/428) N3/T1301 2.01=GETNUM (type 1) 123 => %
10:13:51.968 (390/428) N3/T1301 .001=CallerId: Converted 86353111 => 86353111
10:13:51.984 (390/428) N3/T1301 .000=GETNUM with Calling Number: 86353111
10:13:57.171 (390/428) N3/T632 5.22=CPU Usage (3): 19.53% (HighPri 5.56%, 67us)
10:13:59.531 (390/440) 04:TAM   ICmd 2//1LOGON,TYP=1,AUTOR=+43-1-08               <
10:13:59.609 (390/440) 04:TAM Resp 2101 ATF0022     011122 101359 OK<
10:13:59.640 (390/440) 04:TAM   ICmd 2//2S,N=FXI$86353111,OR=FXC$86353111<
10:13:59.687 (390/440) 04:TAM Resp 2100 OK<
```

In the above example you see that the received DDI number is 123 and the Calling line number is 86353111. Inbound distribution is not done via DDI digits but with the received Calling party number! Within TCFW this reception FAX will look like displayed below:



11. LineServer Trace Options and Hints

11.1 The LS1 Load Balancing Option

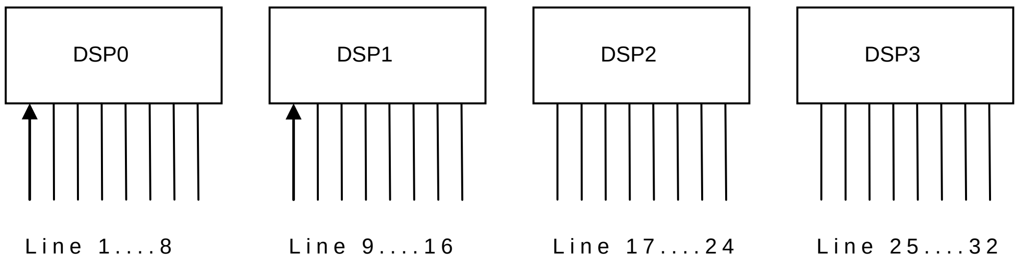
With the introduction of the LineServer hardware, FAX/VOICE communication is handled from so called DSP (Digital Signal processors) modules. Each LineServer cabinet may house up to 4 DSP modules. A DSP module itself is able to handle up to 8 FAX/VOICE communication lines. Every LineServer is able to handle a maximum of 32 FAX/VOICE lines.

With previous releases of the KCS, FAX/VOICE operation always started with the lowest possible channel number available. If the first channel was already busy, a simple counter has been increased by 1 and the next channel has been checked. If this channel was busy too, the next channel has been checked….

With this simple method above it shows clearly what kind of problems you may expect by using a full equipped LineServer. Usually the first DSP module, DSP0, handles lines 1…8, the second handles line 9 … 16, the third 17 … 24 ad the fourth DSP module, DSP3, handles line 25 ... 32.

After the first FAX/VOICE message has arrived at KCS the first channel on DSP0 has been used for reception. During this time a second message arrived, the counter has been increased by one, the second channel on DSP0 has been used and so on. As a worst case, 8 FAX/VOICE calls arrive at KCS at the same time. DSP0 handles all of them while DSP1, 2 and 3 remain in an idle condition.

To prevent this situation and to use every DSP module within a LineServer a so called load balancing option has been implemented. Within the graphic attached below, you see how the load balancing has been implemented:



- DSP0 has already 1 incoming FAX/VOICE call attached
- DSP1 has already 1 incoming FAX/VOICE call attached
- The next incoming FAX/VOICE call will be distributed to DSP2

On the DSP0 which handles FAX lines 1 to 8, one FAX/VOICE call has already arrived. The next FAX/VOICE call has also arrived and has been distributed to DSP1. The next call which will arrive to KCS will be distributed to DSP2.

This load balancing is done from the ISDN module Q.931mux, the multiplexer process which "knows" to which DSP the previous incoming FAX/VOICE call has been distributed.

Load Balancing is switched on or off with position 5 of config line 286. It is switched on in case Bit 7 has been set to 0 (possible values 00, 01, 02 or 03) and it is switched off by setting Bit 7 to 1 (possible values 80, 81, 82, 83).

Please note: By searching for FAX/VOICE problems on specific DSP's you must switch off the load balancing option as otherwise each incoming message is handled from another DSP module.

11.2 The Detection of the LS1 Reference Clock Problem

The LS1 uses a central clock for all connected public ISDN lines. In order to avoid slip errors on the ISDN line it is required that the LS1 derives (synchronizes) its internal clock from the PABX or public switch. A slip error may result in slightly reduced analogue signal quality (sometimes samples are lost or doubled) and therefore mainly XO and XT errors with poor FAX sending quality occurs.

The Lines server derives the reference clock from an active line that is configured as clock slave (position 1 of config line 291 is set to 00 with PRI, BRI line are always clock slave) with a non-zero reference clock priority (position 8 of config line 291):

1) If more ISDN lines are active those with the highest reference clock priority is used

2) If more active ISDN lines have the same clock priority, one of them is selected randomly to be a reference clock

3) If that ISDN being used as reference clock deactivates, another active one (with non-zero reference clock priority) is selected immediately as reference clock line

4) If no reference clock is selected (reference clock priority of all lines is zero), the LS1 is using an internal reference clock that runs asynchronously with the PABX or public exchange. As a consequence a lot of slip errors would occur.

Since TCOSS 7.46.16 this synchronization problem is detected and an error message and event log entry as shown below are created:

`ID:`	`16044`
`TCOSS Level:`	`2 (means Warning)`
`Description:`	`Reference Clock on Line Server %1` **`is not synchronized`** `with %3 on %4.  The current reference clock source is set to %5. This problem may be caused by wrong reference clock priority configuration or defective hardware (TC16/23/24). It may reduce the quality or received and transmitted faxes without error correction mode.`

`Parameters:`	`%1 = location - identifies the Line Server`
	`%3 = Line Type (BRI or PRI)`
	`%4 = Line port (S0a, S0b, S1a or S1b) that is not synchronized`
	`%5 = Current reference clock source`

This error message is repeated once a day and whenever the L1 is re-activated.

If the problem disappears after it has been reported with the error message described above, the following Info message will be created.

```
ID:            16045
TCOSS Level:   1 (means Info)
Description:   Reference Clock on Line Server %1 could be synchronized with
               %3 on %4. The current reference clock source is set to %5.
               This message indicates that a previously detected
               synchronization error does not exist anymore.
Parameters:    %1 = location – identifies the Line Server
               %3 = Line Type (BRI or PRI)
               %4 = Line port (S0a, S0b, S1a or S1b) that is not synchronized
               %5 = Current reference clock source
```

Note: Detection of synchronization problems may take up 10 minutes. Detection of correct reference clock may take up to 20 minutes.

11.3 The LS1 DSP Thread Performance Counter Trace

This trace should only be activated if requested by the development department as it produces a high amount of trace output. This trace shows the performance data of the involved threads of the DSP module before and after a FAX message has been sent and writes the output into the standard TCOSS trace file. It can be used to detect performance problems on any DSP module

This trace output can be activated by changing the following FAX configuration lines (Please note that an additional trace tup channel is required):

```
config line 232 :02 02 02 02,
config line 242 :00 00 00 00 00 00 01 00 00 00 00 00,
```

```
[TCOSS] N3/T107 .015(1lines free)=**** Performance Counter Trace at TickCount=201739ms,
SampleCount=2162354
[TCOSS] N3/T107 .001(1lines free)= 00:Idle             ThTotal:190227ms (190227ms)
IntTotal:1715ms (1715ms) ThreadPure:188512ms (188512ms), Peak=16ms
[TCOSS] N3/T107 .000(1lines free)= 02:.LinkTimer       ThTotal:0ms (0ms) IntTotal:1ms
(1ms) ThreadPure:-1ms (-1ms), Peak=0ms
[TCOSS] N3/T107 .000(1lines free)= 03:.TxLink          ThTotal:205ms (205ms)
IntTotal:4ms (4ms) ThreadPure:201ms (201ms), Peak=1ms
[TCOSS] N3/T107 .000(1lines free)= 04:.trpCtrl         ThTotal:88ms (88ms) IntTotal:0ms
(0ms) ThreadPure:88ms (88ms), Peak=2ms
[TCOSS] N3/T107 .000(1lines free)= 05:.trpCommTa       ThTotal:0ms (0ms) IntTotal:0ms
(0ms) ThreadPure:0ms (0ms), Peak=0ms
```

```
[TCOSS] N3/T107 .035=  TCSD IRQ Time: Min=36us, Max=133us, Max4=113us Total=2161ms
[TCOSS] N3/T107 .090=  TCSD Errors:   BadTimeSlot=0, Skipped IRQs=0
[TCOSS] N3/T107 .060=  Fax Process:   Maxtime=16us, Total=79654us (79654us)
[TCOSS] N3/T107 .005=  Since Last Dump: Time=201739ms IrqTime=2161ms
[TCOSS] N3/T107 .025=  Free TCSD Interrupt Stack: 29568 bytes
[TCOSS] N3/T107 .025=  Free Memory: 2147483647Bytes
[TCOSS] N3/T107 .010=**** End of Performance Counter Trace
```

It might only be interesting for developers and they are the only persons which might request such a trace output. Nevertheless it is mentioned here to give you a clue about the LS1 trace options.

Do only activate this trace for testing purposes or if the development department is asking for it.

11.4 The LS1 Data Pump Trace

This trace output is also called “binary trace”. The binary trace creates a single file for each FAX outgoing or incoming call. These files are stored automatically into the directory `C:\tcoss\trace` on the TCOSS server connected to the LS1.

This **default path** can be changed by setting a new registry value TCOSS\BinaryTrace\Path [REG_SZ]. The maximum **number of files** can be set by registry value TCOSS\BinaryTrace\MaxTraceFiles [REG_DWORD]. The default value is currently set to 50. The **size of the files** can be limited by the registry value TCOSS\BinaryTrace\MaxTraceFileSize [REG_DWORD]. The value defines the maximum size for one file in kilobytes. The default is 10000kBytes

Performance note: The binary trace should only be activated for **one single channel** at a time if PCM trace is used. Since binary trace with PCM produces a large amount of data transfer over the network between the LS1 and the TCOSS server for each channel, activating it with PCM on multiple channels on the same LS1 leads to problems.

Restriction: If binary trace is configured on multiple channels, the difference between the lowest and the highest channel number must be less than 256.

Since binary trace is a whole new feature it requires a different handling than usual trace output. The binary trace produces files with the extension `.btr` in `c:\tcoss\system` on the TCOSS server. These files are binary files and they must be converted for analysis.
Therefore the tool btrc.exe, called “binary trace converter tool, located in directory c:\topcall\lsd\ on the TCOSS server must be used. Under Windows the file extension `.btr` can be associated with that program. If then a binary trace file is double clicked in Windows explorer it gets automatically converted.
By starting the conversion process you will see the following screen output:

```
Binary Trace Converter v1.01.01

Scanning: C:\TCOSS\Trace\TCOSS_000.btr.

Looking for DP messages.
```

```
Creating: C:\TCOSS\Trace\TCOSS_000.asc.

Looking for pcm data.
Detected loss of pcm data, silence inserted!
Creating: C:\TCOSS\Trace\TCOSS_000.wav.

Elements read: 5317. Elements lost: 0. 0.0% loss
Press a key.
```

Depending on the used binary trace level the output of the conversion can be two files: a file with the same name as the binary trace file and the extension **`.asc`** and a file with the extension **`.wav`** which is a standard windows PCM wave file (8 KHz 16 bit stereo).

The **.asc** file is a text file which can be viewed with a simple text editor like Notepad. It contains the messages that are exchanged between TCOSS and the datapump software running on the LS1 depending on the used binary trace level.

The output of such a file might look like displayed below:

```
TCOSS -> Ch41      ** TCOSS to ISDN channel 41
(0:00.010 s) Len:  12, DPMsgId: (1FF) mTXDTMF  , DPMsgPar: 4C 04 C4 09 00 00
00 00 A0 0F 00 00

TCOSS -> Ch41
(0:00.510 s) Len:   0, DPMsgId: (10) mENDTXRX

TCOSS <- Ch41      ** ISDN channel 41 to TCOSS
(0:00.510 s) Len:   0, DPMsgId: (89) mTXEND

TCOSS -> Ch41
(0:00.520 s) Len:   1, DPMsgId: (0B) mCONFIGDMODE   , DPMsgPar: 01
(0:00.520 s) Len:   1, DPMsgId: (0F) mSTARTFAXRX    , DPMsgPar: 0C
```

You see all the messages from and to TCOSS, the time stamps and additionally the channel which has been used for sending/reception, in the example above ISDN channel 41 has been used

This trace output is interesting for the developer only as it requires deeper knowledge of FAX datapump handling and is therefore not further mentioned.

The **.wav** file can be played / viewed with any audio software that supports the standard windows PCM wave format, e.g. the Windows Media Player. Additionally a soundcard and speaker are necessary. It contains the signal, which is sent from LS1 on the right speaker channel and received from the LS1 on the left speaker channel.

The output depends upon the used binary trace level. If only trace level 08 is activated (received PCM data only) the output will be a stereo wave file containing the received signal on the right speaker channel and silence on the left speaker channel.

The recommended trace setting is described below (Please note that an additional trace-tup channel is required):

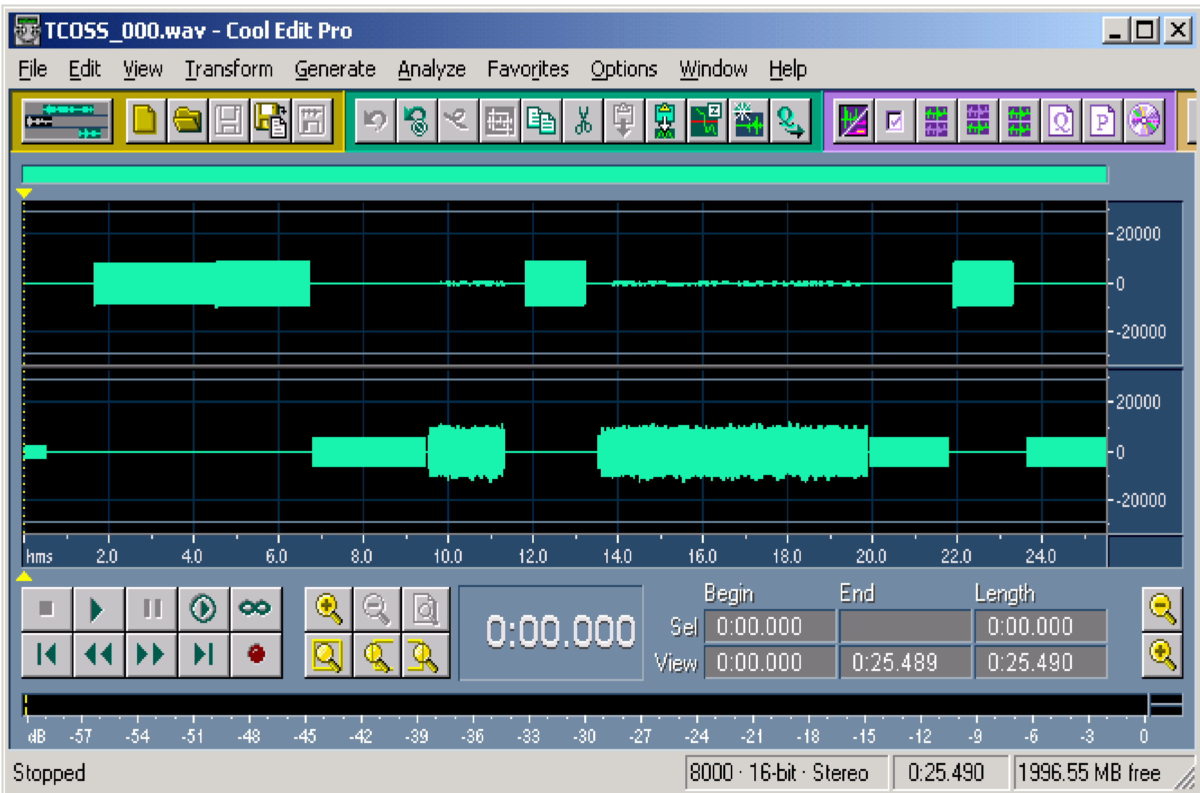
```
config line 232 :02 02 02 02,
```

```
config line 242 :00 00 00 00 0F 03 00 00 00 00 00 00,
```

This trace generates a full sending/reception LS1 datapump trace with additional DTMF information (in case DTMF is used – otherwise empty). If you have activated this trace output and restarted the TCOSS Server, you will see the following information within the standard TCOSS trace file.

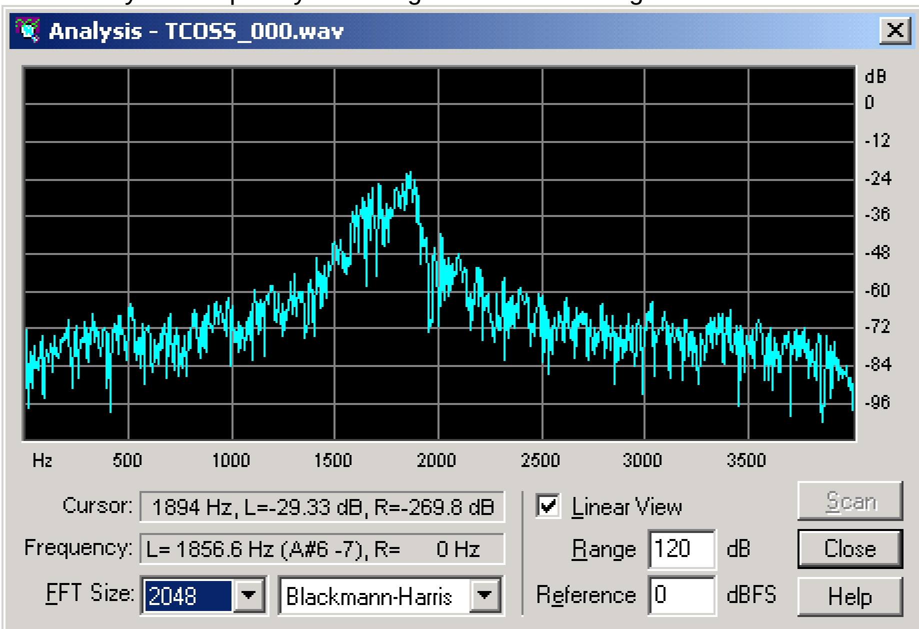
```
[TCOSS] BTR Open File c:\tcoss\trace\TCOSS_000.btr for Channel 41
[TCOSS] BTR Closes File for Channel 41
```

A btr file is created in the standard trace directory which must be converted via the btrc.exe tool, the you can listen to the sent / received FAX message. In case of problems with your LS1 and FAX transmission you should zip the file and attach it to a standard Notes report. The development department is able to further investigate the FAX traffic which special tools (displayed below Cool Edit Pro).



This shows the two channel FAX communication between Sender and Receiver with time stamps.

Additionally the frequency of the signal can be investigated:



Many more investigations or tests are possible, just these two are mentioned here. Nevertheless keep in mind to pack these files in case of problems and attach them to your generated Lotus Notes report.

This would manly help for problems while sending to FAX machines which always fails with a specific FAX transmission error code.

11.5 Using the Same ISDN Line for Incoming FAX/VOICE Calls

With LineServer installations we have the possibility to handle both FAX and VOICE calls via the same physical ISDN line. The only configuration requirement is that voice calls must use separate extensions (extension ranges) and they must be explicitly marked as "VOICE" extensions using the "V" prefix in the ISDN module configuration, lines 254 – 283, like:

```
'11~=V1~ line 254 ,(mark extensions 1000-1999 to be a voice number)
```

The PABX must be setup in the way to route all FAX and VOICE numbers to the same ISDN trunk (BRI or PRI). There are several configuration scenarios on the PABX side how to accomplish this goal:

1) Use the „redirecting number“ feature for incoming VOICE calls

All user’s FAX extensions are routed to the KCS ISDN trunk as usually, and all VOICE calls are handled via one single number reserved for all VOICE calls: the VOICE Access Number (please see separate chapter for details).

2) Using distinct number range for VOICE calls

If the PABX does not support the redirecting number feature, each user will have to be assigned two distinct extensions: one for his FAX and one for his VOICE mailbox.

For example, assume that the number range 1xxx is used for normal VOICE extensions, 2xxx is used for FAX extensions and 3xxx would be VOICE mailbox extensions. The PABX must be setup to route both ranges 2xxx and 3xxx to the same KCS line(s). The VOICE mailbox extensions must be marked as VOICE numbers in the ISDN number conversion config lines:

```
’13~=V3~ line 254, (mark extensions 3xxx to be voice numbers)
```

Further the call forward destination would be different for each user (each user 1xxx would have to setup his call forward destination to 3xxx)

Disadvantage: Each user would in fact occupy 3 different extensions within the PABX.

3) Using different trunk prefixes to distinct number ranges for voice calls

This is very similar with option 2) and may be used in countries without the fix telephone number length (Germany and Austria). For example, assume that the number range 1xxx is used for normal VOICE extensions and the prefix ‚8’ marks FAX extensions (81xxx).

The 1st possibility would be to define another prefix in the PABX for VOICE mailboxes (e.g. ‚89“) and route all such calls to the same KCS line. In addition, the PABX must pass also the trunk prefix to KCS in the dialed number (but attention, by default such trunk „prefixes“ are suppressed by PABX devices and only digits dialed behind the prefix are delivered).

Setup ISDN number conversion table config lines like following:

```
’189~=V~ , (mark extensions 89xxx to be VOICE and remove prefix 89)
’18~=~ , (remove prefix 8 for FAX calls)
```

The call forward destination would be different for each user (each user 1xxx would have to setup his call forward destination to 89xxx).

The 2nd possibility would be to define an additional prefix to be dialed just after the leading „FAX“ prefix („8“), for example „9“, but without changing anything within the PABX. As the PABX by default suppresses the prefix number – in this case „8“ - the PABX would pass only „xxx“ for FAX extensions and „9xxx“ for VOICE mailbox numbers. Setup ISDN number conversion table config lines like displayed below:

```
’19~=V~ , (mark extensions 9xxx to be VOICE and remove prefix 9)
```

The call forward destination would be different for each user (each user 1xxx would have to setup his call forward destination to 89xxx).

11.6 USN – Unique Single Number for FAX/VOICE Calls

This feature only works in combination with a Line Server model 305 (Line Server 1). It does not work with a Dialogic or CISCO telephone engine.

In most of the countries in the world (except for Austria and Germany) the public telephone numbering plan is very restrictive. Telephone numbers have fix length and even local extensions in the companies are the real part of the public numbering plan (e.g. the well known 10-digit numbers in the USA).

If any company needs for example 1000 extensions for its employees it must order (and pay for) 1000 numbers from the local PTT. Therefore companies are often rejecting to order distinct FAX extensions for employees, because it is very cost intensive. They ask for a solution that would make it possible to receive both VOICE and FAX calls for each user via single number/extension.

KCS provides a simple **"entry-level" solution** for this problem, the "Unique Single Number" (USN).

It works as following:

1. Each user receives all FAX and VOICE calls via the same extension. All FAX and VOICE calls are routed by the PABX to his/her telephone at first.
2. ***If the user does not answer the call*** (Cfu, Cfnr or Cfb condition), it is forwarded to KCS server that performs automatic FAX detection and handles the call correspondingly (receives the VOICE mail or the FAX).

   In case the FAX is being sent from a „manual" FAX machine -such a FAX machine does not generate typical fax tones and it cannot be recognized automatically - the caller would hear KCS VOICE menus advising him to press ‚9' to leave a FAX. After he has pressed ‚9' he would immediately hear the KCS FAX prompt and would press the start button on his FAX machine.
3. ***If the user answers the call*** and he hears any fax-like tones, or if the caller would tell him he is just sending a FAX (in the case of „manual" fax machine), he would transfer this call to the single company-wide „FAX Access Number" (this function may be setup to one of telephone's programmable buttons) and hangs off the line. The PABX will at first make a second call to KCS server, and after the user has hung off connect the calling fax with KCS directly.

   This solution is suitable for companies with low fax traffic because once the user have already answered an incoming fax call, he/she will have to transfer it manually to the KCS server's Fax Access Number.

Prerequisites:

1. LS1 server connected via QSIG BRI or PRI trunk to the PABX using the same physical line for both FAX and VOICE calls
2. TCOSS and Voice server must be used
3. PABX must support QSIG call diversion supplementary services („redirecting number") - (For the exact PBX requirements please refer to the „PBX Questionnaire" document)

The reason why QSIG line with call diversion services is required for USN functionality is that this is the only possibility to route all redirected FAX and VOICE calls to the KCS server via one single “Voice Access Number”.

If call diversion service would not work, the USN would not make any sense as distinct FAX and VOICE number ranges would have to be assigned for KCS lines. On the other hand, QSIG protocol is required as it is known to support call diversion services with most of the PABX systems.

Usage of Unique Single number functionality must be configured within the UIF module by changing the third and fourth position of config line 295

:00 00 00 00 00 00 00 00 ,295

Position 4, Automatic FAX detection
00 …. Automatic FAX detection disabled
05 …. Automatic FAX detection enabled (CNG tone detection for 5 seconds

Position 3, Redirecting number for VOICE calls
00 ….Redirecting number for VOICE calls disabled
01 …. Redirecting number for VOICE calls enabled

1. The redirecting number for VOICE calls has to be enabled and the automatic FAX detection has to be set to a value of e.g. 05 (This means that an automatic FAX CNG tone detection is done for 5 seconds)
2. Two distinct extensions must be allocated in the PBX to be routed to the KCS server:

   The „Voice Access Number“ (e.g. „999“, see TC/Voice Access Manual) - Setup this extension as the call forward destination for each user

   The „Fax Access Number“ (e.g. „998“) - Setup a programmable button on each user’s telephone to make a second call to this extension or at least publish this extension in the company’s telephone list so that each user would know that each incoming fax must be transferred to this unique extension
3. Setup the Voice Access Number in UIF module’s number conversion facility to be a Voice number:

   '1999=V999 line 254 ,(mark extension 999 to be a voice number)
4. If the Fax Access Number has been dialed, the incoming fax is routed to the inbox of the user who is originating the fax call transfer – originator based routing):

   '1998=% line 255 ,(mark extension 998 to be a fax access number)
5. Activate the Unique Single Number function for the Voice server (see TC/VoiceAccess documentation).

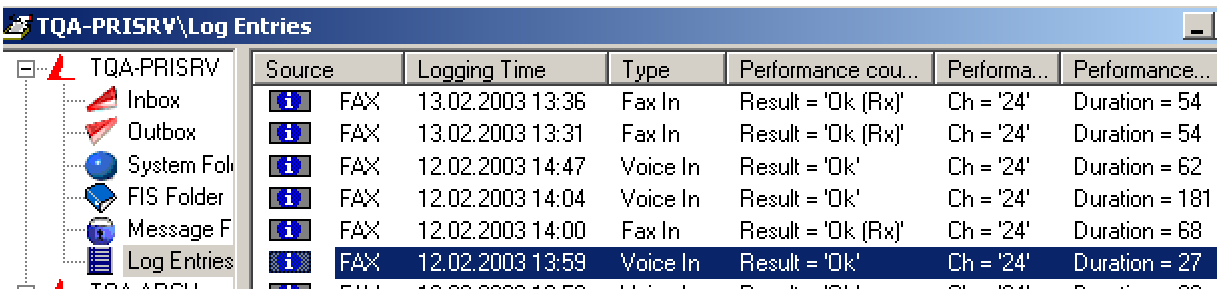
11.7 Logging of incoming FAX/VOICE Calls

With prior TCOSS releases up to 7.47.04, in some cases an incoming FAX/VOICE call did not create a log entry. This behavior caused missing information by using applications like TC/REPORT where statistics about line usage are generated. The situations where no log entries have been created were:

1. An incoming FAX call arrived (e.g. to any FAX inbound number) but no FAX connection could be established. This typically happens if an inbound number is dialed via telephone.
2. If any mailbox command (7x e.g. FIS document retrieval) is used.
3. The call is handled be TC/VoiceAccess

Since TCOSS 7.47.04 it is possible that the FAX module generates user defined log entries. The main reason for using log entries is to provide information that can be used by TC/Report for customer specific reports. But since these log entries can be viewed with TCfW, they may even be used without TC/Report to provide more information about incoming calls for technicians or administrators.

A sample output from TCfW is shown in the screen shot below



The log entry is characterized by the type of incoming call (field “Type”) and its success (field “Result”). The content of these fields can be used as filter to flexible adjust which log entries should be created.

The result value shows basic information about the success of an incoming FAX/VOICE call. Its values are shown in the table below:

Result	Description
Error	Error in command, password, user id, …
No FAX	FAX machine at sender/caller side was expected, but not found.
Ok	A command that does not require communication with a FAX machine executed correctly.
Ok (Tx)	This is the positive response for any **session reversal** command. A connection to the distant FAX machine could be established. The result of the transmission can be found in the fields “FaxEC” and “Pages”.
Ok (Rx)	At least an empty reception document has been created. In that case an entry in the incoming journal (if configured) will be created. The result of the reception can be found in the fields “FaxEC” and “Pages”.

The table below lists all possible types with its possible result values.

Type	Description of type	Result	What happened / Error
FAX In	FAX (with or without inbound number received).	Error	Wrong inbound number
		No FAX	No inbound or inbound number ok, but no sending FAX machine.
		Ok (Rx)	Reception file has been created.
FAX Routing	Receive a FAX and forward it as routing command	Error	Error in destination number
		No FAX	UserId, Password and Command is correct, but no sending FAX machine
		Ok (Rx)	Reception file was created.
FAX Scan	Scanning a document	No FAX	Correct Scan command, but no sending FAX machine
		Ok (Rx)	Reception file was created.
FAX Command	Mailbox commands 71 (without session reversal) or 72	Error	Wrong UserId, Password or command syntax.
		Ok	Successful Mailbox command
FAX Command (SR)	Show Mailbox command (71) with session reversal	No FAX	No FAX machine detected.
		Ok (Tx)	Mailbox content send to connected FAX machine
FIS Command	73xxx command with destination number	Error	Wrong Text number. Error in destination number
		Ok	Send command created
FIS Command (SR)	733xxx command with session reversal	Error	Wrong Text number.
		No FAX	No FAX machine detected
		Ok (Tx)	Text send to connected FAX machine
Voice In	Call handled by TC/VoiceAccess	Ok	Currently no further error information is implemented.

If the Called ID (DDI/DID or DTMF number) contains an 8uupp command the following rules are applied in order to keep privacy of the numerical password.

a) If UserId and password are correct, then the password is replaced by a single “p” character.

b) If UserId or password is not correct, the complete Called ID is set to “wrong user or password”.

If the UserId or Password in a 8uupp command is wrong, it is not possible to interpret the following command. In that case the type “Fax Command” is assumed. This means that e.g. an invalid FIS or routing command will be interpreted as wrong FAX command.

Each log entry may have parameters as shown in the table below.

Field	Type	Description
Source	String	Fixed set to “FAX”
Date/time	Date	End Date/time of connection
Type	String	Type of incoming FAX connection. See extra table above.
Result	String	Basic information about the success of the incoming call. Possible values are described in an extra table above.
Ch	String	TCOSS Channel number
Duration	Integer	Duration of connection in seconds
Called	String	Called DDI/DID/MSN or DTMF number after incoming number conversion.
Redirect	String	Redirecting number (after incoming number conversion). This field appears if the call was redirected by the number “Redirect”. The redirecting number is typically used to forward voice calls to KCS Voice/Access.
Caller	String	Caller number (DDI/DID or DMTF after incoming number conversion) if available.
User	String	User Id from 8uu command in inbound user id. When using inbound FAX in an ASP system, this field is filled with the user Id from the selected storage server.
Reference	String	Name of file that was send/received.
FaxId	String	FAX Called Station Identification (if available)
Pages	Integer	Number successfully received pages or confirmed transmitted pages. If case of reception errors, the additional appended break page is not counted.
FaxEC	String	2 digit FAX error code

The parameters are always created in the sequence shown below, but parameters Duration, Called ID, User, Reference, Pages, FaxId and FaxEC are only optional. If their content is not available or empty it is not stored in the log entry.

The logging function uses the //LOG command. In the very unlikely case that the maximum line length is exceeded, some parameters are discarded without error message.

Additional Notes:

- Log entries are creates by the FAX module (running on TC22 or TC20) after closing the connection. In case of fatal errors (e.g. power failure, shutdown, node reboot, crash..) the log entries for the current active connection are not created.
- In an ASP system all log entries are created on the media server. This allows easier creation of line usage statistics in an ASP system.
- FAX log entries are supported by TC20 based fax and Line Server Model 305 only. They are not supported by TSxx based fax cards
- The fields Caller, Redirect and Called contains the number after conversion through the number conversion table. In case of incoming voice calls the called field will contain the leading “V” character

By default, no logging entries are generated by the FAX module. The content of the “Type” and “Result” field can be used as filter to flexible adjust which log entries should be created. **Note that both filters must match** in order to create a log entry.

The definition of FAX/VOICE call logging - configuration Line 296, is defined as follows

:00 00 00 00 00 00 00 00 ,296

Position 3 **result filter** – the result filter is a set of bits. There is one bit for each possible result value. If the corresponding bit value is set, logging entries with that result value are performed.

Bitvalue: 0 0 0 0 0 0 0 0

Bit 0	filter for result = “Error”
Bit 2	filter for result = “Ok”
Bit 3	filter for result = “Ok (Tx)”
Bit 4	filter for result = “Ok (Rx)”
Bits 5..7	reserved, not used

Position 2 reserved for future types

Position 1 **type filter** – The type filter is a set of bits. There is one bit for each possible type value. If the corresponding bit value is set, logging entries with that result value are performed

Bitvalue: 0 0 0 0 0 0 0 0

Bit 0	filter for type = “FAX In”
Bit 1	filter for type = “FAX routing”
Bit 2	filter for type = “FAX scan”
Bit 3	filter for type = “FAX command”
Bit 4	filter for type = “FAX command (SR)”
Bit 5	filter for type = “FIS command”
Bit 6	filter for type = “FIS command (SR)”
Bit 7	filter for type = “VOICE In”

Value calculation

Bit 7 Decimal Bitvalues Bit 0

128	64	32	16	8	4	2	1

Example: Customer wants to log all successful FIS commands with or without session reversal.

For the FIS command with and without session reversal, Bit 5 and 6 are used. The Bit value for Bit 5 is 32 and for Bit 6, 64. Add these values, 32+64 = 96, convert this decimal value to HEX = 60 HEX. As only successful attempts should be logged, Bit 2 and 3 of position 3 are used. The Bit value for Bit 2 is 4 and for Bit 3 is 8. Add these values, 4 + 8 = 12, convert this decimal value to HEX = 0C. Therefore config line 296 looks like :60 00 0C …

11.8 Supervision of BRI/PRI Line Status

So far (up to TCSP 7.55.12) higher protocol layers of the sending UIF module (for both fax or voice calls) had no knowledge whether the physical line to be used for the particular call is activated or deactivated (out of order or cable disconnected). This fact had a fatal impact especially on the outgoing voice calls that are routed consequently to the idle UIF channel with the highest TCOSS channel number on the particular line server.

So if the highest configured BRI or PRI line was disconnected, no outgoing voice call was possible because all calls attempted to use this disconnected line.

Therefore this problem is solved by a new line layer 1 state supervision mechanism.
Lower protocol layers (BRI/PRI drivers) of all configured UIF modules maintain the current layer 1 status of the connected line and distribute it internally to all connected UIF modules.

This mechanism attempts to keep the line active all the time. If the network side would deactivate the line for any reason, our BRI/PRI driver will attempt to re-activate it immediately because we would not have any other means to recognize whether the line is intentionally deactivated by the network (but cable connected) or the cable is simply not connected.
In the field this may change the behavior only with some BRI point to multipoint lines that are being deactivated by the network after some time of higher layers inactivity. The positive side-effect is that LEDs on BRI interfaces would permanently indicate line active status and thus avoid the confusion why is the line again not active.

The UIF modules receive the current layer 1 status upon following events:

1) Immediately if the line status change was detected (line activation or deactivation)
   This covers the most of BRIs (point to point) and all PRI (E1 and T1 as well) that automatically activate/deactivate upon connecting/disconnecting of a cable

2) Regularly after the layer 1 supervision timer expires (by default 60 seconds, see configuration below).
   This covers more or less only BRI point to multipoint lines that does not activate themselves automatically. Therefore the BRI drivers always attempt to activate the line after the layer 1 supervision timer has expired.

   It means that in a worst case (mainly using of a point to multipoint BRI line) the UIF module may recognize the line activation as late as after 60 seconds.

Configuration

The layer 1 supervision must be configured as follows:

Config line 291, 9th (hex) position:
   3C – the value of layer 1 supervision timer in seconds, default hex 3C (60 seconds).
   00 - layer 1 supervision is deactivated (may only be necessary for some ISDN approvals in order not to confuse the ISDN testing equipment)

   This config position will be automatically set by WConfig on creating new UIF channels or on updating an existing installation up to TCSP 7.55.12

Restrictions

This new layer 1 supervision mechanism is used only for outgoing voice calls, not yet for outgoing fax calls. This means that if one of the lines is disconnected even this line will be used for outgoing fax calls attempts. But this is not so critical as all those failed send orders will go out via one of active lines through send retries later.

11.9 Bit Error Rate Testing (BERT)

11.9.1 How It Works

It is often necessary to verify the quality of the local PBX or PTT interconnection of KCS line servers due to different stability issues (like alarm conditions, slip errors reported by the PBX, or too much of different analog fax errors like XT, XV, XT etc. that may indicate poor line quality).

The easiest method how to verify the quality of the line (even remotely) is to perform so called *Bit Error Rate Testing* (BERT). All what we need is an ISDN test equipment capable of doing BERT testing.

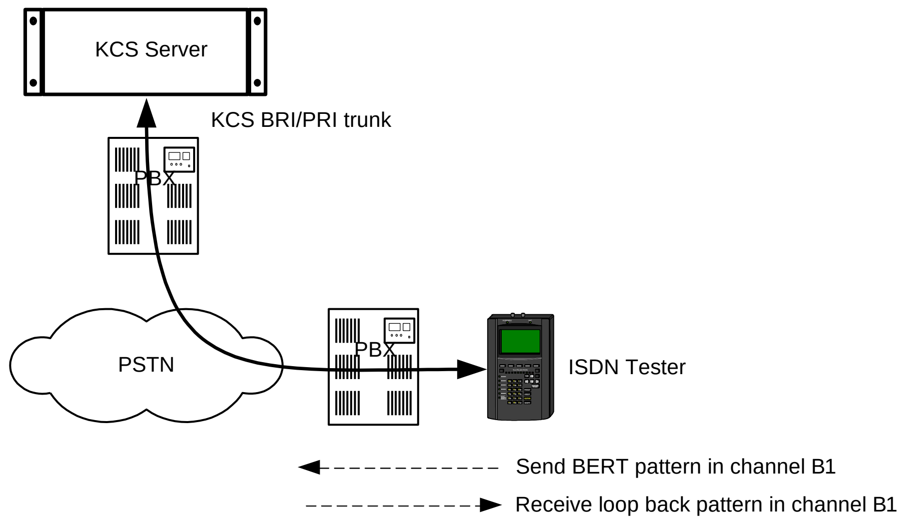
The basic principle is simple: the ISDN tester (Test generator) sends a pseudo-random digital test pattern towards the Device under Test (DUT) that loops the received data transparently back (via the same B-channel). The tester compares the sent and received data for a defined time and makes the test verdict in terms of error ratio (number of bit errors per 1000 bits, for example 1.10^{-6}).

There are three scenarios how to perform BERT:

Scenario 1: Use the ISDN tester as Test generator and the KCS Line server as DUT
(using the B channel Loopback functionality)

The KCS B channel loopback function simply accepts an incoming call to a defined “loopback” extension, and transparently loops the digital data received from the B channel back to the call originator via the same channel.

Once the B channel loopback has already been configured on the customer site, connect the tester to the ISDN line anywhere in the world, call the loopback extension on the customer site and start BERT test procedure. The only condition is the end-to-end digital connection from the tester up to the KCS server across the telephone network:



This test should be seen as an overall test because it verifies the whole connection path between the Tester and the KCS Line Server hardware. If it fails, it should be attempted via another dialup connection. If all of these tests fail, the testing as described in Scenario 2 should be done.

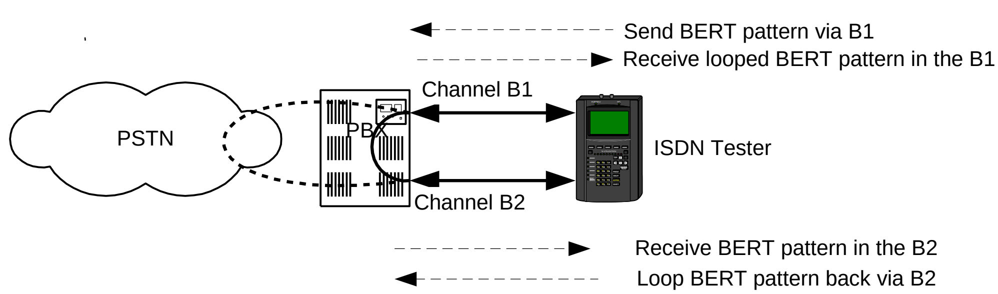
Scenario 2: Use the same ISDN tester as both Test generator and DUT as well

Connect the ISDN Tester to the KCS BRI/PRI PBX trunk and make the following:

1) Make a call to the own number via the PBX (or even via the PSTN public network) for example via the channel B1

2) The PBX routes the call back to the same trunk via different B-channel (for example B2) , answer this call on the Tester and put this into the LOOP mode

3) Start the BERT testing on the 1st connection (channel B1):

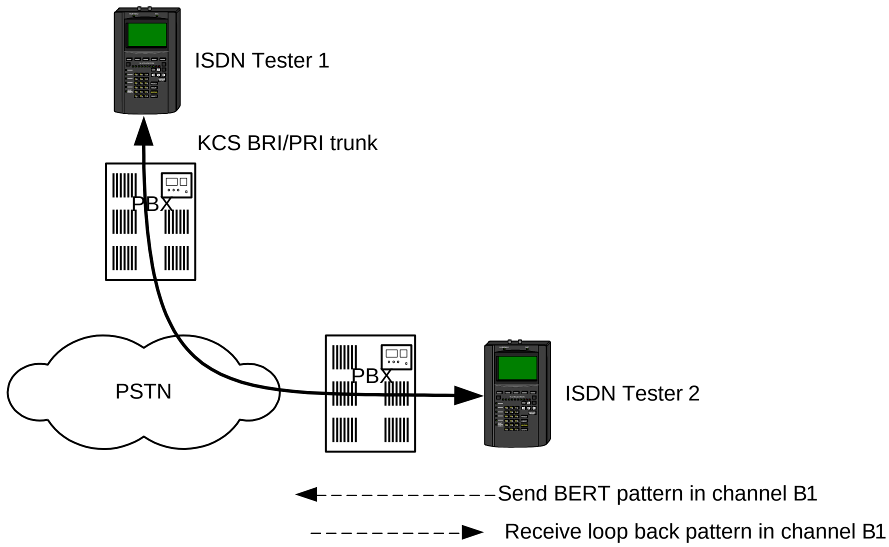


This testing is independent of the KCS hardware and that's why is the best method how to objectively verify the quality of the BRI//PRI trunk and even of the PBX's PSTN interconnection.

Scenario 3: Use two ISDN testers as Test generator and DUT

This scenario is similar to the scenario 1, the only difference is that ISDN tester is being used instead of KCS hardware.

It may be interesting if the scenario 1 testing permanently fails, and the customer owns an ISDN tester capable of performing the LOOP function:



11.9.2 KCS Loopback Configuration

The only configuration need on the KCS side is to allocate a unique incoming extension for the BERT testing and convert it in the number conversion facility (config lines 254-283) in the UIF module into the string “LOOPTEST”. If you are dialing the “LOOPTEST” number you will get a loop test confirmation prompt (1s 880Hz followed by 1s 700Hz). Then the LS1 will echo all received data on this channel without modification. You can try it with a simple telephone and you will hear your own voice with a short delay. But, of cause for real bit error rate tests it will be required to use any ISDN test device (e.g. one of the Aurora family testers) that supports BERT testing).

The BERT test will be terminated if the caller disconnects the call or after a time-out on KCS side. This KCS time-out has a default value of 4000s. If you need any other value, you can specify it in the number conversion table as an integer value immediately after the LOOPTEST. See examples below:

Example of number conversion table:

```
'100=LOOPTEST                           ,line 254-283
'199=LOOPTEST60                         ,
```

If number “00” is called a test loop with the default time-out (=4000s) is activated.

If number “99” is called a test loop with a time-out of 60s (1min) is activated.

Notes:

- During test loop the corresponding channel can’t be used for fax/voice operation. In the lines statistics overview the channel indicates an incoming call during test loop.

- The test loop is not supported with TS29/32/33. With TC20 it is supported with primary rate ISDN lines only! It is fully supported with LS1.
- If you do not have any free DDI/MSN number you can configure a DTMF Prompt and you can define any DTMF number that activates the Loop test. See example below:

Example for loop test activation via extra DTMF prompt:

```
’100=T10                              ,
’24711=LOOPTEST                       ,
```

If you make an incoming call to number "00" you will get a DTMF prompt. If you then enter number "4711" a loop test will be started.

It is a good to allocate and configure a loopback extension during each new installation or upgrade for TCSP 7.50.01 or later even though everything seems to work fine at the moment. But if some stability issues arise later, you can verify the system remotely without any re-configuration.

11.9.3 Performing BERT Tests with Aurora Duet and KCS (Scenario 1)

There is a well-known “Aurora” family of industry-standard ISDN testers (Aurora Duet, Aurora Sonata, Aurora 30) with the capability of performing BERT testing. This chapter describes an example how to make this test using the Aurora Duet tester as Test generator and KCS Line Server as the Device under Test (using of different model of an Aurora tester may require some modification of this procedure):

1) Connect your Aurora Duet to any kind of ISDN line - BRI or PRI, public, local PBX in your office, or on the customer site
   The only prerequisite is that the transmission channel between Aurora and tested KCS must be end-to-end digital (for example, if you would connect Aurora to your local PBX via ISDN line but the PBX itself is connected to the PTT via a/b lines, BERT testing would not work!!!)

2) Make a speech connection to the Loopback extension of the KCS server under the test

3) As soon as the line gets connected, you hear special DTMF prompt (only to identify the loopback function)

4) Keep the Call connected, go to the Connect Menu and press Connect BERT

5) Now Aurora starts sending BERT test patterns towards the LS1 server and compares it with patterns received back.

After the BERT test duration has been reached, a test report will be shown like this (and you can print it out via V.24 cable by going to Results and pressing PRN):

```
RESULT 14 BERT
DATE: 12/06/2003   TIME: 09:59:24
PROTOCOL:           QSIG
CHANNEL:            B1
CPN:                8900
```

```
SUB:
TEST LENGTH:        USER DEF   00:00:10
PATTERN:            USER DEF         2222
THRESHOLD:          UNLIMITED
RESYNC:             25%
ERROR INSERT:       1 in 10^6
ISDN CAUSE          16 LOCAL CLEAR
ELAPSED TIME        00:00:10
BERT RESULT:        PASS                       , general test verdict, PASS or FAIL
TEST TIME:          00:00:10
RX KBITS:           640                        , received data in kBits
BIT ERRS:           0                          , absolute count of bit errors
BER:                0.0E-0                     , Bit error rate
                                  , (e.g.1.0E-3 would mean 1 bit error per 1000 bits)
EFS:                10 (ABS) 100.00%
ES :                0 (ABS) 0.00%
SES:                0 (ABS) 0.00%
US :                0 (ABS) 0.00%
DM :                0 (ABS) 0.00%
S-LOSS:             0
```

Understanding results:

The bit error rate on a digital trunk should not be worse than about 1.0 E-6. On the other hand, this BER testing is an end-to-end test – this means that the full digital path from the tester up to the tested LS1 server will be tested. It means, that the trunk with the worst quality would influence the testing the most (worse result than the KCS-PBX digital trunk itself would produce).

Therefore perform a couple of tests not only one in order to exclude an influence of a bad quality of your connection.

11.9.4 Performing BERT Tests with Single Aurora Duet (Scenario 2)

Is almost the same as testing with Aurora Duett and KCS. The only thing to take into account is that once the incoming call starts ringing on Aurora Duet you should answer it and set it to LOOP mode via Connection menu – Connect LOOP (if Aurora hasn't done it automatically) and then start BERT via the 1st connection.

Please note that Aurora Duet is able to handle two (or more) connections, you will only have to switch between them (1st and 2nd connection) to control them separately (LOOP on the 2nd, BERT on the 1st)

12. KCS Fax Inbound Routing

The KCS system offers several ways for FAX reception. In the following chapters a short overview can be found about the possibilities to receive FAX messages with KCS.

12.1 FAX Inbound Routing via NN99

This is the old method and should not be used anymore. Nevertheless, older installations still use this distribution method. It should be used with a small amount of users only as distribution is done based upon searching the user ID's within a text file which is a slow process.

For this purpose, the FAX channel which is used for reception has to be configured with the used prefix, in our case .FX is used. The FAX configuration might look like below:

```
'.FX  +        ,  235 (only first 4 positions important)
:03 ,236 (Length of DDI info)
```

Within TCFW, the user must have a FAX address which is either marked as "active" or "inactive" and the number must start with a **hyphen**, see below:

Active	No address no.	Service	Number:
X	1	INT	USER A,
	2	FAX	-123,

After you save this FAX address and close the user profiles within TCFW, a question appears whether TCFW should update the KCS system files. Confirm this with yes, and check afterwards the file NN99. You'll find a new entry at the end within the automatically created section of the NN99 as below

```
*,*
FX123,USER A:
```

Inbound FAX message sent to this user will look like attached below within TCFW.

...To Continued	From:	...From Continued	Normalized Sender
.FX123	FAX,+43-1-08	F	F:*43108

12.2 FAX Inbound Routing via rr99

This is the newer and preferred method of inbound routing. It should be used for all installations instead of the NN99 inbound routing. Searching is done via the rr99 system file which is kept in memory. Therefore it is faster than the NN99 text file based routing.

For this purpose, the FAX channel which is used for reception has to be configured with the used prefix, in our case FXI$ is used. The FAX configuration might look like below:

```
'FXI$ +        ,  235 (only first 4 positions important)
:10 ,236 (Length of DDI info)
```

Within TCFW, the user must have an inactive FXI address, see below:

Active	No address no.	Service	Number:
X	1	TOPCALL	USER A,
	2	FXI	123,

After you save this FAX address and close the user profiles within TCFW, check the setup of your system file rr99. It should look like attached below

```
**SENDMODES
**NORMALIZE
**ROUTE
**NODES
**INBOUND
FXI:~,,                fax inbound routing to KCS user
FXI:~,DIST:FXI~,       default fax inbound receiver
```

Inbound FAX message sent to this user will look like attached below within TCFW.

...To Continued	From:	...From Continued	O...	Normalized Sender
FXI,123	FAX,+43-1-08	F		F:*43108

12.3 FAX Inbound Routing via Calling Party Number

This method is also called “originator based routing”. It may be a requirement from your customer to distribute incoming FAX messages based on the “calling party number” and not based upon the DDI number. If the converted received number contains a percent (%) character, then it is replaced by the calling party number.

If no calling party number has been received from the sender, the % character is removed and not used. The percent character has to be inserted via the FAX number conversion table. A possible FAX configuration might look like below:

```
'FXI$          ,  235 (position 1..4 are important)
:20 , 236 (Length of caller ID)
`1~=% , 254 (use calling party number for inbound distribution)
```

Within TCFW, the user must have an inactive FXI address with the expected caller ID. Please note that several different caller ID's can be defined for this single user. A possible configuration might look like

Active	No address no.	Service	Number:
X	1	TOPCALL	USER A,
	2	FXI	43186353111,
	3	FXI	86353131,

After you save this FAX address and close the user profiles within TCFW, check the setup of your system file rr99. It should look like attached below

```
**SENDMODES
**NORMALIZE
**ROUTE
**NODES
**INBOUND
FXI:~,,                fax inbound routing to KCS user
FXI:~,DIST:FXI~,       default fax inbound receiver
```

Inbound FAX message sent to this user will look like attached below within TCFW.

To:	...To Continued	From:	...From Continued	Normalized Sender
USER A	FXI,86353131	FAX,+43-1-08	F	F:*43108
USER A	FXI,43186353111	FAX,+43-1-08	F	F:*43108

If a FAX arrives with a caller ID without matching user profile, the FAX message is either rejected or sent to the distributor queue, depending upon the rr99 definition.

12.3.1 Using Calling Party Number Information Instead of TSI

In the previous examples, the TCFW From: field (originator information) always contained the TSI – transmitted subscriber information as originator.

When receiving faxes on an ISDN line, it is possible the get the Calling Party Number (Caller ID) which provides the number of the calling fax machines. Since this number is verified by PTT network it is protected against manipulation or miss-configuration by the distant user, as it is possible with the TSI.

When using Caller ID, you should be aware about the following restrictions:

- The caller ID may not be available. With ISDN (and Mobile Telephones), the Caller ID may be suppressed by the caller (anonymous call). On analogue lines, it depends on the local network provider (e.g. Austria PPT does not provide caller ID for calls from analogue lines).
- If the distant station uses ISDN with MSN the telephone network ensures that a valid number for this physical line is presented as caller ID. It is the responsibility of the connected devices (e.g. fax machine or PBX) to provide the correct calling party number. If the calling party number is not correct or missing, the telephone network may use the main telephone number instead.
- If the distant line uses ISDN with DDI the telephone network ensures that the telephone number of the line (without extension) is correct. The correct extension must be provided by the distant PBX. It is not verified or changed by telephone network.
- The Caller ID is not supported with 1TR6 protocol

A possible FAX configuration might look like below:

'FXI$ +   FXC$, 235 (position 11..14 are important)

If you would receive a FAX message with DDI information and caller ID, the FAX is distributed using the DDI information to search for the user ID which belongs to that user, and the From: field (originator field) contains the calling party number instead of the TSI, see below:

...To Continued	From:	...From Continued	O...	Normalized Sender
FXI,123	FXC,43186353111	F		F:43186353111

13. Appendix A – Troubleshooting ISDN

Verifying ISDN connections may be divided into three main steps:

1) Check the ISDN line (E1/T1 with primary rate, S/T interface with basic rate)

With ISDN PRI start TCOSS and check both LEDs on the back-plane of TC34: if the line is OK, only the GREEN LED must be ON. If not, the line is not working properly. Follow the advices from following Troubleshooting table.

With ISDN BRI, there are no line status indications on the TC33/TS33 interfaces. Try to make an outgoing send attempt. If you get an error code, follow up the advices from the troubleshooting table.

2) Check the outgoing connections

Once the E1/T1 line (with PRI) is working properly, you may try to send a FAX. Consider that processing of the ISDN send order consists of two subsequent steps:

1.) The ISDN connection to the distant fax machine is being established
2.) The conventional analogue fax G3 transmission occurs via established ISDN connection

If there was a problem during ISDN connection establishment, the send order breaks with the ISDN error code Ix (IA, IC, …) and in the author field of the send order the more detailed error explanation can be found:

ISDN Info nnnn,

where the 'nnnn' is the ISDN error cause value returned from the network side (PTT or PABX), or generated locally by KCS due some specific error situations.

Please check this cause and find out in the Troubleshooting Table below what to do.

If the ISDN send order breaks with any of analogue FAX error codes like XL,XF … the ISDN connection has already been established, but the subsequent fax transmission failed for some reason. Please refer to the Troubleshooting table below for explanation.

3) Check the incoming connections

a.) If the ISDN line was subscribed for both incoming and outgoing calls, it is a good idea to try to send a FAX via the ISDN line to the own number so that the network side (PTT or PABX) would route the call back into KCS via the same line. If getting any problems during this “loop” test (your outgoing send order breaks with any error) follow up the outgoing part of the Troubleshooting table. But please note that although this “loop” test works with the most of configurations, there may be situations where sending in a loop is not permitted at network side by some reason and you will get corresponding ISDN Info code. With PABX connections, you may try the “loop” test internally and also externally via PTT. Thus the “loop” test is a very good first test step but when it fails it does not necessarily mean the KCS system malfunction.

b.) Having a pure incoming KCS system take an analogue or ISDN phone set and try to dial into the KCS system. When everything works, you should hear the fax prompt for at least several seconds. Now you could proceed to send an incoming fax from other manual fax machine available.
If you could not hear the fax prompt, please follow up the incoming part of the ISDN Troubleshooting table for the solutions.

Getting the dialed number on incoming calls

All supported ISDN protocols are able to deliver either the whole dialed recipient’s number or at least its variable part (extension) to the KCS. This number is used by KCS inbound routing mechanism to route incoming faxes to KCS recipients.

The corresponding ISDN feature with point to point lines is in Europe being referred to as DDI (Direct Dial In). On the other hand, with ISDN networks in the USA and Japan this feature is being referred to as “Delivery of Dialed Number”.

Both methods (European, US and Japanese) are handled in the same way by the KCS system: simply by activating DDI through configuration.

The similar feature with point to multi-point lines is being referred to as “MSN” (Multiple Subscriber Number). If you activate MSN via configuration, you may also route incoming faxes according to the MSN numbers.

The common problem on configuring DDI/Delivery of Dialed Number/MSN is that you sometimes will not know the length of the dialed recipient’s number you will get from the network. Basically the network may provide you with the whole dialed number (“base” number + variable extension) or only variable extension. Try at first to get this information from the network administrator. If you don’t succeed follow up the Troubleshooting table how to learn the length of delivered number.

ISDN Troubleshooting table

The ISDN Troubleshooting table describes the most of typical error situations that may occur in the field and proposes the solution that should cure the problem. It can be used for all supported ISDN protocols supported except for 1TR6.

It is universal for both primary rate (PRI) and basic rate (BRI) interfaces as the most of error situations/error codes are the same. Those proposed solutions that make sense only with PRI interface are marked as "only for PRI" (checking E1/T1 line status, B-channels configuration etc.)

The first column denotes 3 important facts:

1) Direction: Outgoing call, Incoming call or Both

2) Error class: The general nature of the error situation, there are following error classes:

- Line problem (E1/T1 with PRO or S/T with BRI)
- Normal error (e.g. user busy, wrong dialed number etc.)
- Resource unavailable (at the network side)
- Service or option not available (at the network side)
- Service or option not implemented (at the network side)
- Invalid message (may indicate serious ISDN protocol problem)
- Protocol error (may indicate serious ISDN protocol problem)
- Interworking (network problem during internetworking with another transit networks)
- Miscellaneous
- analogue fax errors (error during outgoing fax transmission)
- incoming errors

3) Error source: which side of the user-network interface reported the problem

Direction Error class Error source	Problem	Short explanation	Solution
Line problem Error source: KCS	TC34 LEDs: Only RED is ON	Line alarm condition And no signal on the line	Please see "Verifying E1/T1 line" (only with PRI).
	TC34 LEDs: RED and GREEN ON	Line alarm condition Any signal on the line	Please see "Verifying E1/T1 line" (only with PRI).
	ISDN Info 202	Line not activated	Please see "Verifying E1/T1 line" (only with PRI). Verify the configuration (especially lines 250 and 286) Let the network administrator verify the network interface (reset, activate etc.) Change the hardware (TC34-only with PRI, or TC33/TS33 with BRI).
	ISDN Info 200	Layer 2 problem	Line active, but layer 2 could not be established. Verify layer 2 config parameters, config line 289, especially pos. 1 (user/network side). Try to change this setting (user->network or network->user) Let the network administrator verify whether layer 2 is active.

Outgoing call: Normal error Error source: Network	ISDN Info 1	unallocated (unassigned) number	The dialed number is wrong, or currently not assigned. Try another one.
	ISDN Info 2	No route to specified transit network	Either wrong number or a network problem to reach the destination. Try another number and report the problem to the network administrator.
	ISDN Info 3	No route to destination	
	ISDN Info 6	Channel unacceptable	The B-channel selected for this call is unacceptable for the network. Verify (only with PRI): • Which B channels are active on the line and whether you properly configured them ('A'/'B'-side, bitwise B-channel map config line 287, pos. 1-4). • Whether you did not configure more UIF modules than the number of active B-channels with this line. If this error persist for longer time (and your configuration is ok), the network side may hang. Try to reconnect the line and restart the TC20/TC34 interface. Report the problem to the network administrator.
	ISDN Info 7	Call awarded and being delivered in an established channel	Should not normally occur. If it occurs very often, indicates an ISDN protocol problem. Make an ISDN trace and report the problem to Kofax
	ISDN Info 16	Normal call clearing	The call has been cleared most probably by the destination user. This error may rarely occur during normal operation, but may indicate a problem: when the network or destination user want to clear the call, they should use more detailed error code. If this error occurs very often verify: • If it occurs only with specific number, the problem lies by destination user himself or by his supporting network. • If all send orders break with this error it indicates probable ISDN protocol problem on network side Report the problem to the network administrator.
	ISDN Info 17	User busy	The destination number is busy, try another one.
	ISDN Info 18	No user responding	The user does not respond to the call, problem on the destination side.
	ISDN Info 19	No answer from user (user alerted)	
	ISDN Info 21	Call rejected	The destination user does not wish to accept the call even though he could have accepted it because he is neither busy nor incompatible with fax G3. The KCS system may also return this error when: • Just after TCOSS start (equipment not yet ready) • During normal operation when it becomes not ready for receiving faxes (e.g. not configured for incoming operation, disk full)
	ISDN Info 22	Number changed	The dialed number is wrong, or currently not assigned, most probably it has changed. Try another one.
	ISDN Info 26	Non-selected user clearing	Should not occur normally. If so, make an ISDN trace and report the problem to Kofax.
	ISDN Info 27	Destination out of order	The destination equipment is not working properly, e.g. switched off, cable disconnected etc. This error is reported by the network serving the destination equipment. Try another number.
	ISDN Info 28	Invalid number format	The dialed number is wrong, most probably incomplete. Try another one. Verify whether you have properly configured numbering plan used (conf. line 286, pos. 6). In Japan you need "unknown", with all other protocols "ISDN/Telephony numbering plan".

Outgoing call	ISDN Info 29	Facility rejected	The requested facility cannot be provided by the network, verify: • Whether the network (PABX) supports bearer capability "3.1 kHz" or at least "Speech", consult with network administrator • Whether the network (PABX) supports unrestricted ("clear") 64 KBit circuit mode B-channels, consult with network administrator • ISDN services (config line 251)-all possibilities • With T1 line and AT&T protocol verify whether the proper AT&T service has been provisioned for outgoing calls. If the service has been provisioned on call-by-call basis, you must configure it in the config line 286. • A-law/u-law configuration (conf. line 250, pos. 5 (A-law used in Europe, u-law in the USA and Japan) • Which B channels are active on the line and whether you properly configured them ('A'/'B'-side, bitwise B-channel map config line 287, pos. 1-4, only for PRI) • Whether the line was configured for fax G3 (voice-band data). • With PABX try both external and internal sending. If internal works and external not, you maybe have no rights for external sending • With BRI and Euro-ISDN protocol, try conf. line 286 pos. 3 to 01. Report the problem to the network administrator.
	ISDN Info 30	response to status enquiry	Should not normally occur. If it occurs very often, indicates an ISDN protocol problem. Make an ISDN trace and report the problem to Kofax
	ISDN Info 31	Normal, unspecified	The same as ISDN Info 16
	ISDN Info 34	No circuit/channel available	The B-channel selected for this call is not available. Verify (only with PRI) : • Which B channels are active on the line and whether you properly configured them ('A'/'B'-side, bitwise B-channel map config line 287, pos. 1-4) • Whether you did not configure more UIF modules than the number of active B-channels with this line. This error may: • Occur temporarily due to the call collision problem. If it occurs too often with PRI, see "handling B-channel assignment collision". • Persist for a longer time on network hang. Try to reconnect the line or restart KCS system. Report the problem to the network administrator
Resource unavailable	ISDN Info 38	Network out of order	Network side is not functioning correctly and this condition is likely to last for a longer period of time. If it occurs very often, report the problem to the network administrator.
Error source: Network	ISDN Info 41	Temporary failure	Network side is not functioning correctly and this condition is not likely to last for a longer period of time. It may occur very rarely during normal operation. If it occurs very often, it may indicate the equipment (PABX) congestion. Report the problem to the network administrator.
	ISDN Info 42	Switching equipment congestion	The switching equipment (PABX) is experiencing a period of high traffic. It may occur rarely during normal operation. If it occurs very often, report the problem to the network administrator.
	ISDN Info 43	Access information discarded	Some access information could not be delivered to the destination user. As KCS does not use any kind of user-to-user signaling, it should not occur normally. If so, make an ISDN trace and report the problem to Kofax.
	ISDN Info 44	Requested circuit/channel not available	The same as ISDN Info 34
	ISDN Info 47	Resources unavailable, unspecified	Self-explanatory, perform similar actions as recommended with ISDN Info 29 and 34.
Outgoing call:	ISDN Info 49	Quality of service unavailable	Similar to ISDN Info 29, see actions there. There seems to be an authorization problem. With PABX, especially check both external and
Service or option not available	ISDN Info 50	Requested facility not subscribed	internal sending. Report the problem to the network administrator
Error source:	ISDN Info 57	Bearer capability not	

Network		authorized	
	ISDN Info 58	Bearer capability not presently available	
	ISDN Info 63	Service or option not available, unspecified	
Outgoing call:	ISDN Info 65	Bearer capability not implemented	Verify whether network side supports “3.1kHz” audio or at least “Speech” bearer capability, consult it with network administrator. For possible actions see also ISDN Info 29.
Service or option not implemented	ISDN Info 66	Channel type not implemented	Verify whether the network provides for unrestricted (“clear”) 64 KBit circuit mode B-channels, consult with network administrator. For possible actions see also ISDN Info 6 and 29.
Error source: Network	ISDN Info 69	Requested facility not implemented	See ISDN Info 29
	ISDN Info 70	Only restricted digital information bearer capability is available	Request unrestricted (“clear”) 64 KBit circuit mode B-channels from the network administrator.
	ISDN Info 79	Service or option not implemented, unspecified	See ISDN Info 29
Outgoing call:	ISDN Info 81	Invalid call reference value	Improper configured call reference length (config line 286, pos. 1) • PRI interface needs CR length 2 • QSIG protocol needs always CR length 2 (also with BRI!) • BRI needs CR length 1 (except for QSIG)!
Invalid message	ISDN Info 82	Identified channel does not exist	The B-channel selected was not activated for the line at the network side. See ISDN Info 34.
Error source:	ISDN Info 83	A suspended call exists	Should normally not occur, as the KCS does not support call
Network	ISDN Info 84	Call identity in use	suspend/resume functionality. If so, make an ISDN trace and report the
	ISDN Info 85	No call suspended	problem to Kofax.
	ISDN Info 86	Call having the requested call identity had been cleared	
	ISDN Info 88	Incompatible destination	The destination equipment is not compatible. Check and extend the ISDN services config line 251. Probably you had only fax G3 service configured. Add also service without any HLC indicator. Try another number (this error should be destination number dependant). If it persists with all numbers, it indicates a problem on the network (PABX side).
	ISDN Info 91	Invalid transit network selection	Should not occur as KCS does not indicate any transit network. If so, make an ISDN trace and report to Kofax.
	ISDN Info 95	Invalid message, unspecified	See ISDN Info 96.
Outgoing call:	ISDN Info 96	Mandatory information element is missing	Network side claims that one of KCS ISDN messages misses one of the mandatory parts or even does not recognize KCS messages themselves. This error indicates a serious ISDN protocol problem. Verify the configuration: • especially config lines 250 and 286 • With BRI and Euro-ISDN protocol, try conf. line 286 pos. 3 to 01 If the problem persists, make an ISDN trace and report to Kofax.
Protocol error	ISDN Info 97	Message type non-existent or not implemented	
Error source: Network	ISDN Info 98	Message not compatible with call state of message type non-existent or not implemented	
	ISDN Info 99	Info Element non-existent or not implemented	
	ISDN Info 100	Invalid information element contents	
	ISDN Info 101	Message not compatible with call state	
	ISDN Info 102	Recovery on timer expire	Our outgoing send attempt (SETUP message) is ignored by the network/PABX side. • ISDN service compatibility: try another ISDN services (config line 251)

			• Verify the configuration, especially lines 250 and 286. • Try sending to another number (local and external). • With T1 line and AT&T protocol verify whether the proper AT&T service has been provisioned for outgoing calls. If the service has been provisioned on call-by-call basis, you must configure it in the config line 286. • Please make an ISDN trace and report the problem to Kofax
	ISDN Info 111	protocol error, unspecified	See ISDN Info 96.
Outgoing call: Internetworking Error source: Network	ISDN Info 127	Internetworking, unspecified	There was an internetworking problem with a network that does not support more precise error cause values. Try another destination number. Report the problem to the network administrator.
Outgoing call:	ISDN Info 128	cause info element with zero length received	Should normally not occur. The call was cancelled with an invalid cause information. Make an ISDN trace and report to Kofax.
Miscellaneous	ISDN Info 129	no cause value (=ISDN info code) has been received	Should normally not occur. The call was cancelled without any cause information. Make an ISDN trace and report to Kofax.
Error source: KCS	ISDN Info 201	Restart procedure invoked by the network during the call	The network side invoked RESTART procedures during the call establishment phase. This may occur rarely during normal operation. If it occurs too often, it may indicate an ISDN protocol problem. Make an ISDN trace and report to Kofax.
	ISDN Info 300	Connect request timeout	Destination user alerted but hasn't accepted the call before the connect request timer expired (config line 284). There are several possibilities: • The destination number was a telephone number and no one picked up the phone. Verify the number. • ISDN problem occurs at destination side (PABX). Try to extend ISDN services config line 251 and to increase the connect request timer config line 284.
	ISDN Info 301	Call collision in Layer 4	The send attempt was cancelled due to the call collision (the incoming fax is handled with priority). May occur during normal operation.
Outgoing: Analogue FAX errors Error source: KCS	Fax error XL	No answer from distant telefax	ISDN connection has already been established, but there is no fax machine at the destination number.
	Fax errors X0,XA	Call collision in fax module	The send attempt was cancelled due to the call collision (the incoming fax is handled with priority). May occur during normal operation.
	Other FAX errors	Problem during fax transmission	Please refer to TCOSS Ssystem Manual, UTF errors description
Incoming Error source: Unknown (Network or KCS)	KCS does not respond to incoming call	You hear ring tone until network fails to timeout and breaks the call	Most probably the network does not route the incoming call to KCS. Disconnect the ISDN cable from KCS and try again: • If you now immediately hear the busy or number not available tone, the call has really been routed to KCS, but its ISDN interface seems not to work properly. Try to reboot it. It the problem persists after reboot or it occurs too often, it may indicate a serious KCS problem. Please make an ISDN trace and report it to Kofax. • If you still hear the ring tone for a longer time, the call has really not been routed to KCS. Report to network administrator.

	KCS responds with busy tone to phone call	KCS not ready/not configured for incoming operation	• All KCS lines are busy with other incoming calls. Wait for several minutes and try again. • You have tried the incoming call just after TCOSS start up and UIF modules may not yet be ready for incoming calls. Wait for several minutes and try again. • KCS rejects the call because its UIF module(s) is not configured for incoming operation. Check the configuration with config program. • You work with MSN but don't have any own number(s) configured with conf. lines 254-283. To accept all MSN numbers you may set conf. line 254 to '1~=~ • KCS cannot accept incoming calls due to disk full condition. • KCS rejects the call as it believes it comes from an incompatible equipment. Try another telephone set/fax machine. Make an ISDN trace and report to Kofax.
	KCS responds with 3 short beeps followed by busy tone to phone call	DDI/MSN misinterpreted as fax command (send attempt from another KCS breaks with XL or XU error)	The received DDI/MSN is longer than max. DDI/MSN length conf. by line 236. Maybe the network sends complete dialed number instead of only the extension. See DDI/MSN not working properly section.
	KCS responds with fax prompt to phone call but incoming fax doesn't work	DDI/MSN not working properly (send attempt from another KCS breaks with XN error)	• You use DDI/MSN (conf. line 252 is '2' or '1') but get DDI/MSN with different number of digits than expected (some networks may send the whole dialed number and some only the variable extension as DDI/MSN) • You have invalid inbound prefix (conf. line 235) • NN99 or rr99 does not route the call to existing KCS user Verify whether the problem lies really in DDI/MSN: • Switch DDI/MSN off (conf. Line 252 to '0'), clear the inbound prefix (conf. line 235) and try the incoming fax. If now incoming fax is accepted, the problem lies in DDI/MSN. If not, KCS may have a problem with this fax equipment. Try another one or if possible send an incoming fax from another KCS. If config without DDI/MSN works verify what kind of DDI/MSN you get: • Switch DDI or MSN on again (conf. line 252 to '2' or '1') • Delete conf. lines 254-283 and set line 254 to '1~=~ • Set max. inbound number length conf. line 236 to 14 (hex) • Set proper inbound routing via rr99: - define service FXI with prefix FXI (as configured in conf. line 235) - enter line FXI:~,DIST:~ into the INBOUND section of rr99 - create KCS user DIST - send an incoming fax with proper DDI/MSN Check the inbox of DIST user for the dialed number: • There is only prefix FXI, but no additional digits. You don't get any DDI/MSN/dialed number from network. This feature is probably not activated on the network side. Report the problem to network administrator. • There is prefix "FXI" and some additional digits that came as DDI/MSN/dialed number from the network. Adjust your DDI/MSN configuration to support provided DDI/MSN length (conf. lines 254-283, 235, 236, rr99 or NN99). Refer to the TCOSS System Manual.
	Incoming fax accepted but not properly routed	Inbound routing not working	You use DDI/MSN (conf. line 252 is '2' or '1') but:: • don't get any DDI/MSN digits from the network. DDI/MSN must be activated on the network side. Contact network administrator. • Wrong NN99/rr99: all incoming faxes routed to default recipient You have forgotten to conf. DDI/MSN (conf. line 252 to '2' or '1')

14. Appendix B – FAX Communication Errors

Layer 7 errors

		(1)	(2)
X0	call collision in layer 7	1	1
X1	attempt to send an empty document	5	5
X2	error when opening back-received file	5	5
X3	error in back reception	5	5
X5	error during reception	y)	y)

Layer 6 errors

XA	call collision in layer 6	1	1
XB	data error within TCI-block	5	5
XC	form buffer is out of memory	5	5

Layer 2 errors

XG	receiver not ready timeout	4	4
XF	no answer from the distant station	2	1
XH	line occupied by local telefax unit	1	1
XI	error in selection number	5	5
XJ	no dial tone	2	1
XK	answer back mismatch	5	5
XL	no answer from distant telefax	5	1
XM	illegal identification of called station	4	1
XN	illegal response during training phase	4	1
XO	three learn attempts unsuccessful	4	1
XP	illegal page confirmation	4	1
XQ	page transmitted incorrectly	4	4
XR	illegal frame received	4	1
XS	unable to find appropriate baud rate	4	1
XT	no response received	4	1
XU	busy or no dial tone on PBX connection	2	1
XV	unexpected end of document	y)	y)
XW	too many line distortions at training sequence	y)	y)
XX	illegal identification received	y)	y)
XY	no command received	y)	y)
XZ	illegal command received	y)	y)

(1) …. break code for transmission

(2) …. break code for extended dialing mode (switch x)

y) …. error code for reception only

15. Index

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