Spojovací soustavy

přednáška č.5.

Studijní podklady k předmětu „ Spojovací soustavy “ pro studenty katedry elektroniky a telekomunikační techniky

Obsah

8. Signalizace

8.1. Druhy signalizace v systémech 1. a 2. generace

U systémů 1. a 2. generace se vnitřní signalizace, která se přenáší mezi řídícími částmi ústředny, označuje též jako signalizace **mezistupňová**. Signalizace přenášená mezi ústřednami se označuje jako signalizace **síťová**. U spojovacích systémů 2. generace se mezistupňová a síťová signalizace dělí na

- registrovou signalizaci
- linkovou signalizaci

8.1.1. linková signalizace

- mezi spojovacími systémy na různých spojovacích vedeních (dvoudrátové, čtyřdrátové, ...), dohlíží na spojení po celou dobu jeho trvání

8.1.2. registrová signalizace

- uplatňuje se při výstavbě spojovací cesty, kdy jsou v činnosti spolupracující registry a příslušné kódové přijímače-vysílače, nejčastěji se používá multifrekvenční mfc R2

rozdělení dle místa přenosu:

- vnitřní (mezistupňová)
- síťová
- účastnická

8.2. Linková signalizace

signalizace typu P

- vychází ze stejnosměrné signalizace spojovacího systému P 51, ze strany synchronních systémů se pracuje stejnosměrnými impulsy, včetně přenosu volby a tarifikačních impulsů, u asynchronních systémů přenos účastnické volby přebírá MFC registrová signalizace R2. Jednotlivé řídící signály jsou vyjádřeny různými potenciály na a, b drátech a doplněny stavy na c drátu.

signalizace typu I

- u impulsní signalizace jsou dopředné a zpětné značky vysílány třemi různými délkami impulsů, modulovanými signálem o kmitočtu např. 50 Hz (dle typu přenosového systému), krátký impuls (40až85)ms, střední impuls (250až350)ms, dlouhý impuls (950až2000)ms.

- dvoudrátové okruhy – 50 Hz
- čtyřdrátové okruhy – 2280 Hz

signalizace typu T

- trvalou signalizaci je možné použít pouze na čtyřdrátových okruzích, v dopředném i ve zpětném směru jsou trvale vysílány řídící signály a to po jednom dvoudrátovém okruhu v dopředném směru a po druhém ve zpětném , na kmitočtu např. 3 825 Hz (mimo hovorové pásmo), v průběhu výstavby spojovací cesty je doplněna signalizací MFC R2.

signalizace typu K

Používá se pro PCM 30/32 nejčastěji doplněná o MFC-R2 (K+MFC-R2),v využívá 2. signalizační kanály (bity a,b) v 16. KI pro každý směr přenosu. Při digitálním spojení do ústředen bez možnosti MFC-R2 se vysílá volba impulsně (K+DEC), DEC=dekadika.

- signalizace K je signalizace typu CAS (Channel Associated Signalling), tzn. pro každý hovorový kanál má rezervovaný a pevně přidělený signalizační kanál pro linkovou signalizaci.
- laskavý čtenář promine několik poznámek k PCM 30/32 pro zopakování: KI = time slot neboli časová pozice kanálu v rámci, 32 kanálových intervalů po osmi bitech tvoří rámec, šestnáct rámců vytváří multirámec, v jednom rámci je 30 hovorových kanálů a dva služební (0.KI a 16.KI), 0.KI přenáší synchronizační značky rámcové synchronizace, 16.KI přenáší značky linkové signalizace, v ostatních KI se přenáší značky registrové signalizace (pokud se použije) a hovorové vzorky, 15 po sobě jdoucích rámců přenese v 16 KI linkovou signalizaci pro všech 30 hovorových kanálů, první přenášený rámec multirámce obsahuje v 16 KI multirámcovou synchronizační značku 00001011 - začátek multirámce, fyzické parametry digitálního rozhraní dle ITU-T G.703 a G.704.
- pro každý hovorový kanál jsou rezervovány v 16 KI příslušného rámce 4 bity abcd, zatímco v některých zemích se používají až tři bity v české variantě se používají dva bity ab, bit c je trvale nastaven c=0 a bit d=1.
- pro sestavení spojení je nutné přenášet informace o čísle volaného a k tomu se nejlépe pro signalizaci K hodí registrová signalizace MFC-R2 (doporučení ITU-T Q.421-Q.424)
- signalizace K se provozuje ve variantách K+DEC (dekadická volba), K+MFC (registrová MFC-R2), K+DTMF (používá se pouze ve vyjímečných případech K/U převodníků = pokud je použit MUX, který převádí signalizaci K na účastnickou sig. U, ústředna je tedy připojena na analogové rozhraní se smyčkovou signalizací, např. připojení na P51, může být dokonce obousměrné!!!).V národní verzi se obě varianty (K+DEC, K+MFC) používají zasadně jako

jednosměrné, přičemž se PCM trakt může rozdělit na příchozí a odchozí kanály, obvykle 15/15.

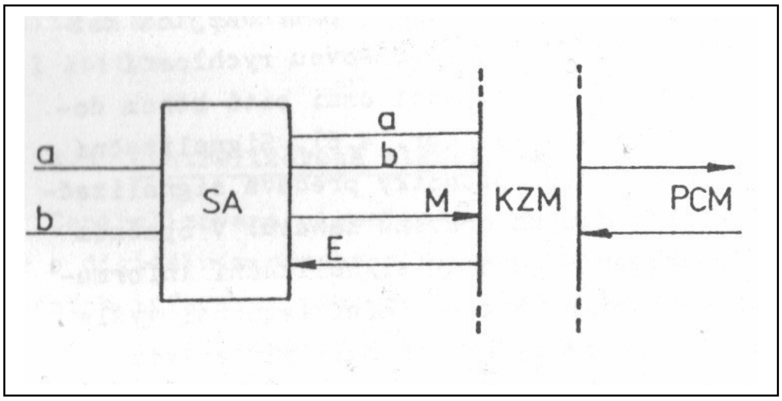
- přehled značek K signalizace (d-dopoředná, z-zpětná), neuvádím napojení, flash a blokaci:

klid	10 z	
obsazení	00 d	
potvrzení obsazení	11 z	
impuls volby	01 d	60 ms
mezera volby	00 d	40 ms
mezičíslicová mezera	00 d	800-1000 ms
přihlášení	01 z	600 ms
tarifní impulz	00 z	100 ms
zachycení	10 z	
závěr	11 z	
vybavení	10 d	
zpětné vybavení	00 z	

- tarifní impulz 00 po 100 ms přejde do mezery mezi tarifními imp. 01, což je stav přihlášení a zůstává do příchodu dalšího tarif. impulzu

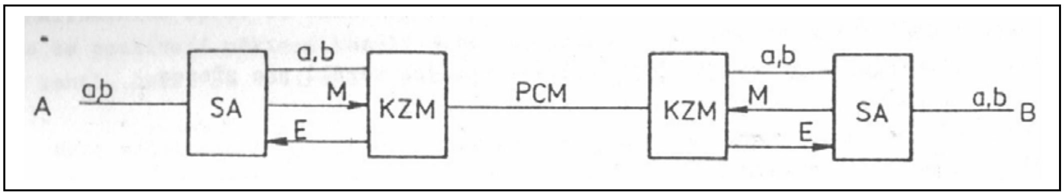
Princip přeměny analogově vyjádřených řídících signálů na digitální a naopak

Signalizace z anal. systémů 1. a 2. generace přichází po a,b drátech do A/D převodníku, pro zpracování signalizace se používá signalizační adaptor SA, který přijímá anal. řídící signály a přeměňuje je na impulsy vyjádřené zemním potenciálem, vysílají se po signalizačním drátě M do koncového zařízení multiplexu KZM, kde se přemění na digitální tvar a zařadí do multiplexu PCM.

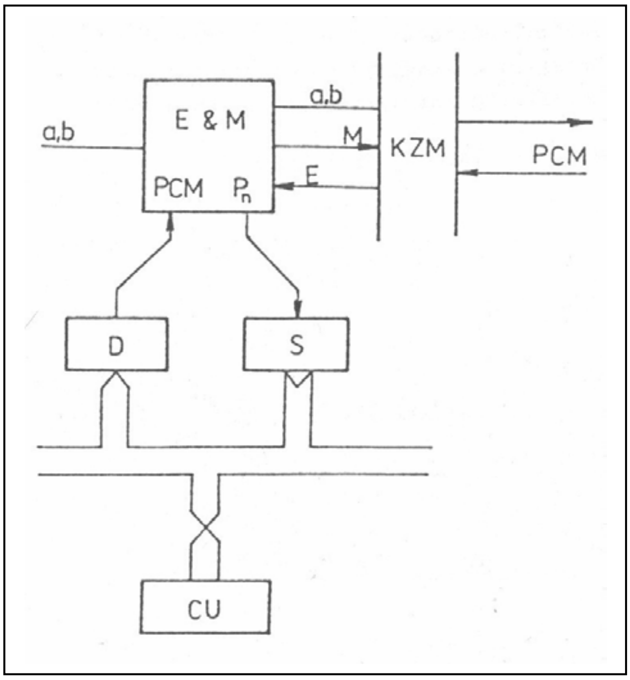


Řídící přijaté z PCM v digitálním tvaru se v KZM přemění na impulsy vejádřené zemním potenciálem a předávají se po signalizačním drátě E do signalizačního adaptoru, odtud se v analogově vyjádřenými signály předávají po drátech a,b do analogové ústředny.

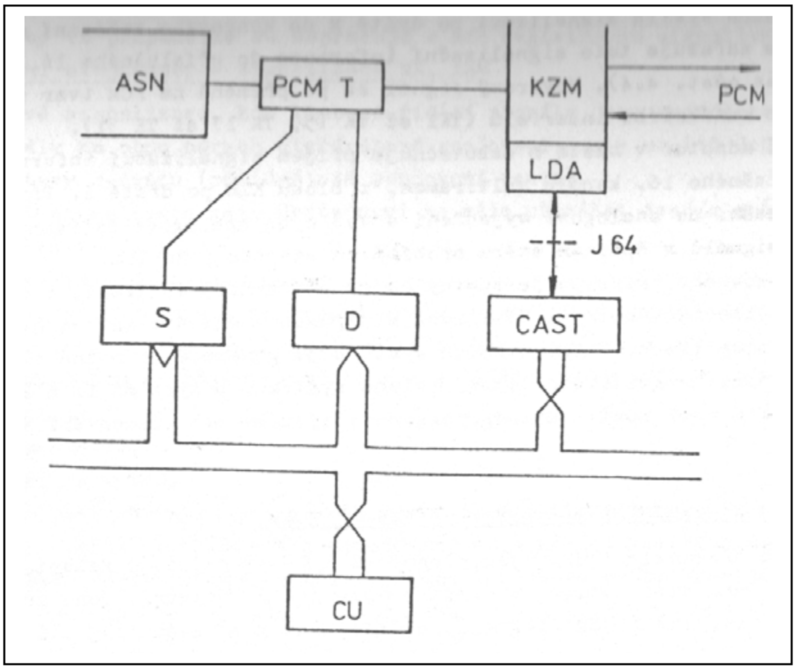
První princip řešení představuje použití 30-ti SA, z nichž vychází 30 E&M drátů vstupujících do KZM, digitálně vyjádřená informace se zařazuje do kanálového intervalu KI 16 multiplexu PCM, hovorové signály jsou zařazovány do příslušných KI 1 až 15 a KI 17 až 31.



Efektivnější způsob spočívá v SW řešení zpracování signalizace, indikování signálů na drátě E a generování signálů na drátě M se uskutečňuje pomocí snímače S (Scanner) a rozdělovače D (Distributor), oba prvky jsou řízeny jednotkou CU (Control Unit), nevýhodou zůstává přeměna z digitálního tvaru prostřednictvím E&M na analogový a naopak. Toto řešení se označuje jako zpracování signalizace pomocí *bloku elektronických přenášečů E&M*.



Při plně digitálním zpracování musí být každý blok KZM vybaven vlastním datovým adaptorem LDA (Local Data Adaptor). Tento adaptor je přes rozhraní J 64 propojen se signalizačním adaptorem . Zde se přenáší řídící informace mezi KZM a řídící jednotkou CU. Metoda plně digitálního zpracování má výhodu v tom, že signalizační informace nejsou konvertovány prostřednictvím E&M signalizace. Signalizační informace jsou přenášeny mezi KI 16 a řídící jednotkou CU, signalizace se směruje na datový adaptor LDA, osmibitová slova se vysílají rychlostí 64 kbit/s do signalizačního terminálu CAST (Channel Associated Signaling Terminal). CAST předává signalizační informace včetně jejich příslušnosti k určitému hovorovému kanálu do řídící jednotky CU. V opačném směru CAST vytváří obsah 16 KI včetně multirámcového souběhu.



8.3. Registrová signalizace MFC-R2

Pro dopředné značky se používají skupiny I a II, pro zpětné skupiny A a B, nejdříve si registry vyměňují značky skupiny I se skupinou A, po přechodu (A-3) se přechází na skupinu II se skupinou B. Jedna z užitečných vlastností je přenesení čísla volajícího (*ANI - Automatic Number Identification*), používá se k tomu opakovaná značka A-5, proces zachycení identifikace volajícího může být vyvolán kdykoliv během spojení, v některých zemích (např. Finsko) nedojde k přihlášení pokud není volající identifikován. Funkce ANI používána zváště u digitálních PBX, které si vyžádají identifikaci volajícího během kódové výměny a zobrazují toto číslo na displejích digitálních přístrojů nebo zapisují do databází pro příchozí tarifikaci, funkce ANI u PBX je v ČR k nemilé radosti Telecomu velice oblíbená a veškeré počáteční snahy Telecomu znemožnit identifikaci volajícího z veřejné sítě vyšly naprázdno, protože ústředny SEL 100 S12 a EWSD se naštěstí řídí doporučením ITU-T .

- značky MFC-R2 registrové signalizace - uvedu pouze nejpoužívanější:

I -1 až I -10	číslice 1 až 0
I - 12	ANI není možné
I - 15	konec ANI, konec volby
II - 1	úč. nebo AT (spojovatelka) bez možnosti napojení
II - 2	úč. s předností (možnost čekání ve frontě)
II - 5	národní AT s možností napojení
A - 1	vysílat číslici n+1
A - 3	adresa neúplná, přechod B-značky a vysílat kategorii volajícího
A - 4	neprůchodnost
A - 5	vysílat kategorii volajícího či vysílat ANI po předchozí A-5
A - 6	adresa úplná, propojit hovorovou cestu a po přihlášení tarifovat
B - 3	úč. obsazen, napojení možné
B - 4	neprůchodnost
B - 5	neexistující úč.
B - 6	úč. volný, tarifovat
B - 7	úč. volný, netarifovat

8.4. Signalizace v digitálních systémech a sítích

Signalizace na PCM 30/32

CAS – kanálově orientovaná signalizace (přiřazena hovorovým kanálům) – jde o signalizaci K doplněnou o MFC-R2, volba se přenáší v hovorovém Kl, zatímco u K+DEC je volba přenášena v 16 Kl.

CCS – signalizace společným kanálem – jedná se o centralizovaný přenos signalizačních zpráv po společných kanálech,

8.5. Signální systém CCITT č.6

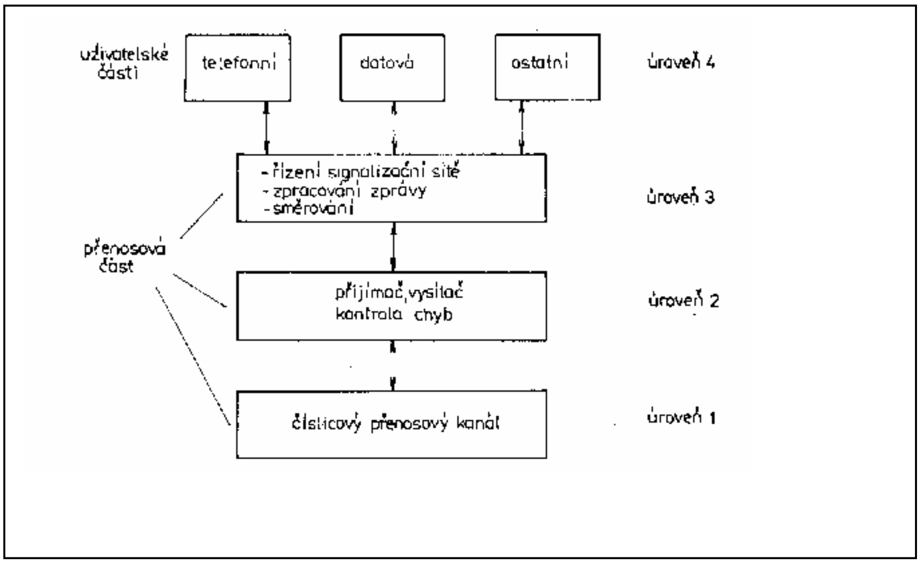
- všechny potřebné informace pro spolupráci dvou ústředen jsou přenášeny ve společném signalizačním kanále, signalizace se přenáší po zvláštním signalizačním kanále obousměrně rychlostí 2 400bit/s

- přenášená zpráva nese informaci o okruhu, ke kterému patří, umožňuje sledování provozu, lokalizaci poruch a závad, vytváření provozních statistik (jednotlivé zprávy mají konstantní délku 28 bitů)

- SS CCITT č.6 byl původně určen pro analogové kanály, byl však časem přizpůsoben i pro přenos po digitálních kanálech (ve třech variantách), pro digitální telekomunikační sítě byla CCITT signalizace č.7

8.6. Signalizace č.7.

- využívá se společný digitální kanál (vyhrazený s přenosovou rychlostí 64 kbit/s nebo celé ISDN PRI toky), je předpokladem k zavedení ISDN, kde každá služba je samostatným uživatelem signalizace. Vychází z modelu OSI (Open System Interconnection), který popisuje sedm vrstev. V signalizaci č.7. nám postačí definovat čtyři úrovně (fyzická, linková, síťová a uživatelská)



layer - physical, link, net, user

- fyzická vrstva – je nejblíže přenosovému prostředí, zajišťuje příjem/vysílání zpráv v kódu HDB3, multiplexace/demulůtiplexace signalizačních zpráv
- linková (spojovací) – zabezpečuje přenos, detekce/korekce chyb, kontrola chybovosti
- síťová – zpracování signalizačních zpráv, řízení signalizační sítě

koncepce vychází z rozdělení signalizace na dvě části:

- MTP (Message Transfer Part), přenosová část MTP1 až MTP3 – přenos signalizačních zpráv
- UP (User Part), uživatelská část – zpracování signalizačních zpráv

Formáty signalizačních zpráv

- formát zprávy mezi uživatelskou a přenosovou částí (mezi 3. a 4. vrstvou)

zpráva	adresa			význak služby
	okruhu	cílový SP	zdrojový SP	
8.n	12	14	14	8

- formát zprávy mezi fyzickou a linkovou vrstvou (1. a 2.)

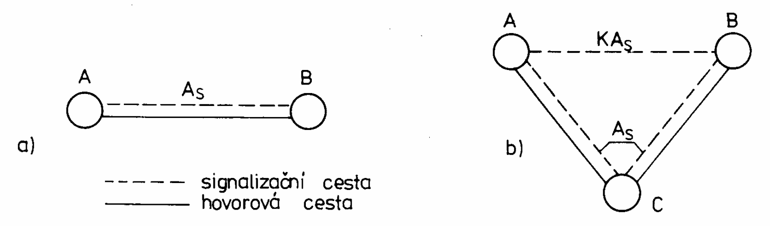
synchr.	kontrola	zpráva	adresa	význak	indik. délky	pořadí	potvrz.	synchr.
8	16	8.n	40	8	8	8	8	8
		zpráva z UP						

Asociativní přenos signalizace:

signalizační a hovorová cesta je stejná

Kvaziasociativní přenos signalizace:

signalizační a hovorová cesta je různá



Signalizační zprávy SS7

IAM – Initial Address Message

SAM – Subsequent Address Message (nese číslice při volbě)

ACM – Address Complete Message (informace o volbě je úplná)

ANM – ANswer Message (přihlášení)

CPG – Call ProGress (před přihlášením)

REL – RELease

RLC – ReLease Completed

Adresace má tři části:

OPC – originating point code

DPC – destination point code

SLS - signaling link selection

8.7. Signaling System 7 - SS7R2

Definition and Overview

Signaling System 7 (SS7) is an architecture for performing out-of-band signaling in support of the call-establishment, billing, routing, and information-exchange functions of the public switched telephone network (PSTN). It identifies functions to be performed by a signaling-system network and a protocol to enable their performance. The signaling architecture SS7 defines a completely new and separate signaling network. The network is built out of the following three essential components, interconnected by signaling links:

signal switching points (SSPs)—SSPs are telephone switches (end offices or tandems) equipped with SS7-capable software and terminating signaling links. They generally originate, terminate, or switch calls.

signal transfer points (STPs)—STPs are the packet switches of the SS7 network. They receive and route incoming signaling messages towards the proper destination. They also perform specialized routing functions.

signal control points (SCPs)—SCPs are databases that provide information necessary for advanced call-processing capabilities.

Once deployed, the availability of SS7 network is critical to call processing. Unless SSPs can exchange signaling, they cannot complete any interswitch calls. For this reason, the SS7 network is built using a highly redundant architecture. Each individual element also must meet exacting requirements for availability. Finally, protocol has been defined between interconnected elements to facilitate the routing of signaling traffic around any difficulties that may arise in the signaling network.

To enable signaling network architectures to be easily communicated and understood, a standard set of symbols was adopted for depicting SS7 networks. Figure 2 shows the symbols that are used to depict these three key elements of any SS7 network.

Figure 2. Signaling Network Elements

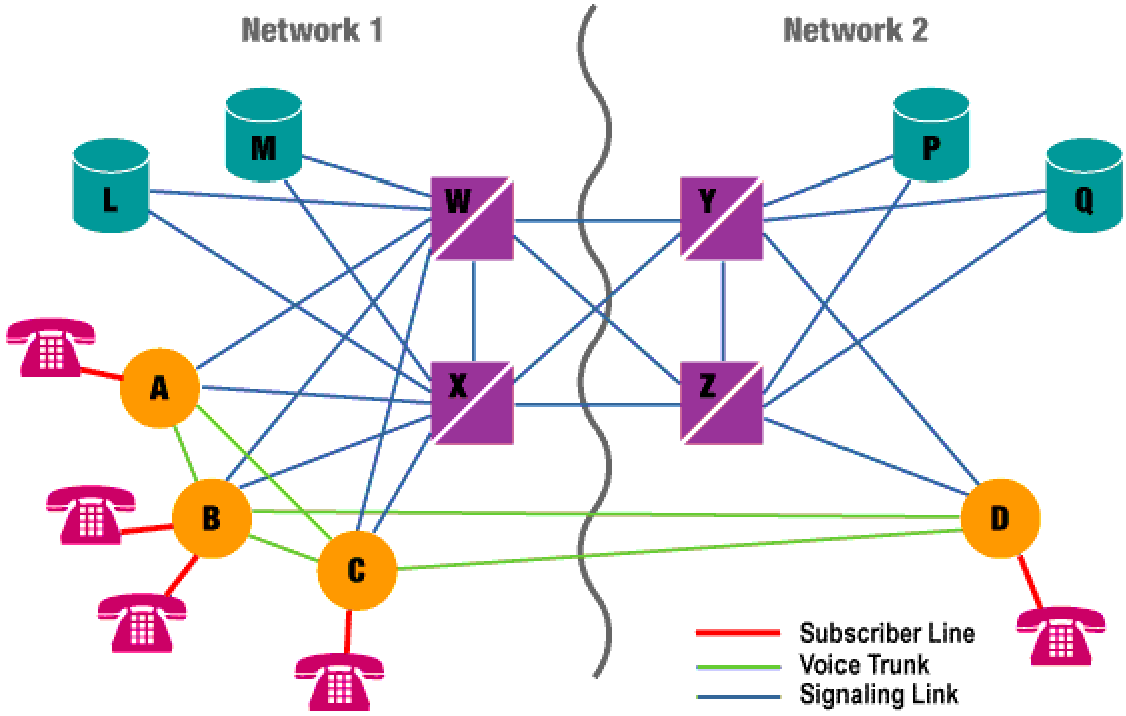
STPs and SCPs are customarily deployed in pairs. While elements of a pair are not generally co-located, they work redundantly to perform the same logical function. When drawing complex network diagrams, these pairs may be depicted as a single element for simplicity, as shown in Figure 3.

Figure 3. STP and SCP Pairs

Basic Signaling Architecture

Figure 4 shows a small example of how the basic elements of an SS7 network are deployed to form two interconnected networks.

Figure 4. Sample Network



The following points should be noted:

1. STPs W and X perform identical functions. They are redundant. Together, they are referred to as a mated pair of STPs. Similarly, STPs Y and Z form a mated pair.

2. Each SSP has two links (or sets of links), one to each STP of a mated pair. All SS7 signaling to the rest of the world is sent out over these links. Because the STPs of a mated pair are redundant, messages sent over either link (to either STP) will be treated equivalently.

3. The STPs of a mated pair are joined by a link (or set of links).

4. Two mated pairs of STPs are interconnected by four links (or sets of links). These links are referred to as a quad.

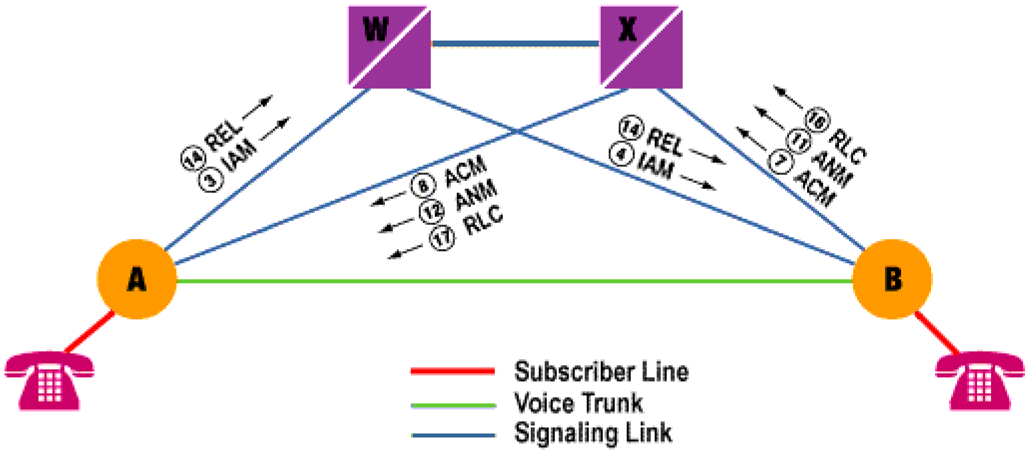
5. SCPs are usually (though not always) deployed in pairs. As with STPs, the SCPs of a pair are intended to function identically. Pairs of SCPs are also referred to as mated pairs of SCPs. Note that they are not directly joined by a pair of links.

6. Signaling architectures such as this, which provide indirect signaling paths between network elements, are referred to as providing quasi-associated signaling.

Call Setup Example

Before going into much more detail, it might be helpful to look at several basic calls and the way in which they use SS7 signaling (see *Figure 6*).

Figure 6. Call Setup Example



In this example, a subscriber on switch A places a call to a subscriber on switch B.

1. Switch A analyzes the dialed digits and determines that it needs to send the call to switch B.

2. Switch A selects an idle trunk between itself and switch B and formulates an initial address message (IAM), the basic message necessary to initiate a call. The IAM is addressed to switch B. It identifies the initiating switch (switch A), the destination switch (switch B), the trunk selected, the calling and called numbers, as well as other information beyond the scope of this example.

3. Switch A picks one of its A links (e.g., AW) and transmits the message over the link for routing to switch B.

4. STP W receives a message, inspects its routing label, and determines that it is to be routed to switch B. It transmits the message on link BW.

5. Switch B receives the message. On analyzing the message, it determines that it serves the called number and that the called number is idle.

6. Switch B formulates an address complete message (ACM), which indicates that the IAM has reached its proper destination. The message identifies the recipient switch (A), the sending switch (B), and the selected trunk.

7. Switch B picks one of its A links (e.g., BX) and transmits the ACM over the link for routing to switch A. At the same time, it completes the call path in the backwards direction (towards switch A), sends a ringing tone over that trunk towards switch A, and rings the line of the called subscriber.

8. STP X receives the message, inspects its routing label, and determines that it is to be routed to switch A. It transmits the message on link AX.

9. On receiving the ACM, switch A connects the calling subscriber line to the selected trunk in the backwards direction (so that the caller can hear the ringing sent by switch B).

10. When the called subscriber picks up the phone, switch B formulates an answer message (ANM), identifying the intended recipient switch (A), the sending switch (B), and the selected trunk.

11. Switch B selects the same A link it used to transmit the ACM (link BX) and sends the ANM. By this time, the trunk also must be connected to the called line in both directions (to allow conversation).

12. STP X recognizes that the ANM is addressed to switch A and forwards it over link AX.

13. Switch A ensures that the calling subscriber is connected to the outgoing trunk (in both directions) and that conversation can take place.

14. If the calling subscriber hangs up first (following the conversation), switch A will generate a release message (REL) addressed to switch B, identifying the trunk associated with the call. It sends the message on link AW.

15. STP W receives the REL, determines that it is addressed to switch B, and forwards it using link WB.

16. Switch B receives the REL, disconnects the trunk from the subscriber line, returns the trunk to idle status, generates a release complete message (RLC) addressed back to switch A, and transmits it on link BX. The RLC identifies the trunk used to carry the call.

17. STP X receives the RLC, determines that it is addressed to switch A, and forwards it over link AX.

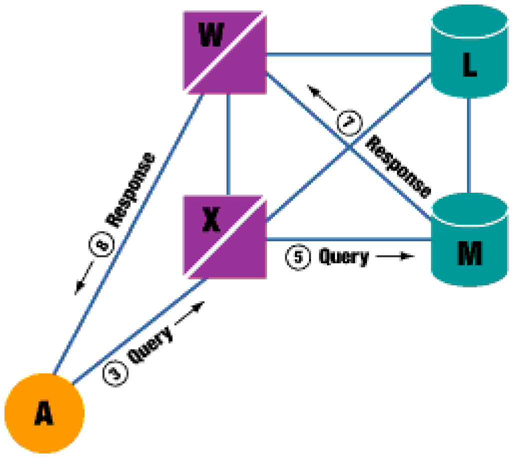
18. On receiving the RLC, switch A idles the identified trunk.

Database Query Example

People generally are familiar with the toll-free aspect of 800 (or 888) numbers, but these numbers have significant additional capabilities made possible by the SS7 network. 800 numbers are virtual telephone numbers. Although they are used to point to real telephone numbers, they are not assigned to the subscriber line itself.

When a subscriber dials an 800 number, it is a signal to the switch to suspend the call and seek further instructions from a database. The database will provide either a real phone number to which the call should be directed, or it will identify another network (e.g., a long-distance carrier) to which the call should be routed for further processing. While the response from the database could be the same for every call (as, for example, if you have a personal 800 number), it can be made to vary based on the calling number, the time of day, the day of the week, or a number of other factors. The following example shows how an 800 call is routed (see *Figure 7*).

Figure 7. Database Query Example



1. A subscriber served by switch A wants to reserve a rental car at a company's nearest location. She dials the company's advertised 800 number.

2. When the subscriber has finished dialing, switch A recognizes that this is an 800 call and that it requires assistance to handle it properly.

3. Switch A formulates an 800 query message including the calling and called number and forwards it to either of its STPs (e.g., X) over its A link to that STP (AX).

4. STP X determines that the received query is an 800 query and selects a database suitable to respond to the query (e.g., M).

5. STP X forwards the query to SCP M over the appropriate A link (MX). SCP M receives the query, extracts the passed information, and (based on its stored records) selects either a real telephone number or a network (or both) to which the call should be routed.

6. SCP M formulates a response message with the information necessary to properly process the call, addresses it to switch A, picks an STP and an A link to use (e.g., MW), and routes the response.

7. STP W receives the response message, recognizes that it is addressed to switch A, and routes it to A over AW.

8. Switch A receives the response and uses the information to determine where the call should be routed. It then picks a trunk to that destination, generates an IAM, and proceeds (as it did in the previous example) to set up the call.

Layers of the SS7 Protocol

As the call-flow examples show, the SS7 network is an interconnected set of network elements that is used to exchange messages in support of telecommunications functions. The SS7 protocol is designed to both facilitate these functions and to maintain the network over which they are provided. Like most modern protocols, the SS7 protocol is layered.

Physical Layer

This defines the physical and electrical characteristics of the signaling links of the SS7 network. Signaling links utilize DS–0 channels and carry raw signaling data at a rate of 56 kbps or 64 kbps (56 kbps is the more common implementation).

Message Transfer Part—Level 2

The level 2 portion of the message transfer part (MTP Level 2) provides link-layer functionality. It ensures that the two end points of a signaling link can reliably exchange signaling messages. It incorporates such capabilities as error checking, flow control, and sequence checking.

Message Transfer Part—Level 3

The level 3 portion of the message transfer part (MTP Level 3) extends the functionality provided by MTP level 2 to provide network layer functionality. It ensures that messages can be delivered between signaling points across the SS7 network regardless of whether they are directly connected. It includes such capabilities as node addressing, routing, alternate routing, and congestion control.

Collectively, MTP levels 2 and 3 are referred to as the message transfer part (MTP).

Signaling Connection Control Part

The signaling connection control part (SCCP) provides two major functions that are lacking in the MTP. The first of these is the capability to address applications within a signaling point. The MTP can only receive and deliver messages from a node as a whole; it does not deal with software applications within a node.

While MTP network-management messages and basic call-setup messages are addressed to a node as a whole, other messages are used by separate applications (referred to as subsystems) within a node. Examples of subsystems are 800 call processing, calling-card processing, advanced intelligent network (AIN), and custom local-area signaling services (CLASS) services (e.g., repeat dialing and call return). The SCCP allows these subsystems to be addressed explicitly.

The second function provided by the SCCP is the ability to perform incremental routing using a capability called global title translation (GTT). GTT frees originating signaling points from the burden of having to know every potential destination to which they might have to route a message. A switch can originate a query, for example, and address it to an STP along with a request for GTT. The receiving STP can then examine a portion of the message, make a determination as to where the message should be routed, and then route it.

For example, calling-card queries (used to verify that a call can be properly billed to a calling card) must be routed to an SCP designated by the company that issued the calling card. Rather than maintaining a nationwide database of where such queries should be routed (based on the calling-card number), switches generate queries addressed to their local STPs, which, using GTT, select the correct destination to which the message should be routed. Note that there is no magic here; STPs must maintain a database that enables them to determine where a query should be routed. GTT effectively centralizes the problem and places it in a node (the STP) that has been designed to perform this function.

In performing GTT, an STP does not need to know the exact final destination of a message. It can, instead, perform intermediate GTT, in which it uses its tables to find another STP further along the route to the destination. That STP, in turn, can perform final GTT, routing the message to its actual destination.

Intermediate GTT minimizes the need for STPs to maintain extensive information about nodes that are far removed from them. GTT also is used at the STP to share load among mated SCPs in both normal and failure scenarios. In these instances, when messages arrive at an STP for final GTT and routing to a database, the STP can select from among available redundant SCPs. It can select an SCP on either a priority basis (referred to as primary backup) or so as to equalize the load across all available SCPs (referred to as load sharing).

ISDN User Part (ISUP)

ISUP user part defines the messages and protocol used in the establishment and tear down of voice and data calls over the public switched network (PSN), and to manage the trunk network on which they rely. Despite its name, ISUP is used for both ISDN and non–ISDN calls. In the North American version of SS7, ISUP messages rely exclusively on MTP to transport messages between concerned nodes.

Transaction Capabilities Application Part (TCAP)

TCAP defines the messages and protocol used to communicate between applications (deployed as subsystems) in nodes. It is used for database services such as calling card, 800, and AIN as well as switch-to-switch services including repeat dialing and call return. Because TCAP messages must be delivered to individual applications within the nodes they address, they use the SCCP for transport.

Operations, Maintenance, and Administration Part (OMAP)

OMAP defines messages and protocol designed to assist administrators of the SS7 network. To date, the most fully developed and deployed of these capabilities are procedures for validating network routing tables and for diagnosing link troubles. OMAP includes messages that use both the MTP and SCCP for routing.

What Goes Over the Signaling Link

Signaling information is passed over the signaling link in messages, which are called signal units (SUs).

Three types of SUs are defined in the SS7 protocol.

message signal units (MSUs)

link status signal units (LSSUs)

fill-in signal units (FISUs)

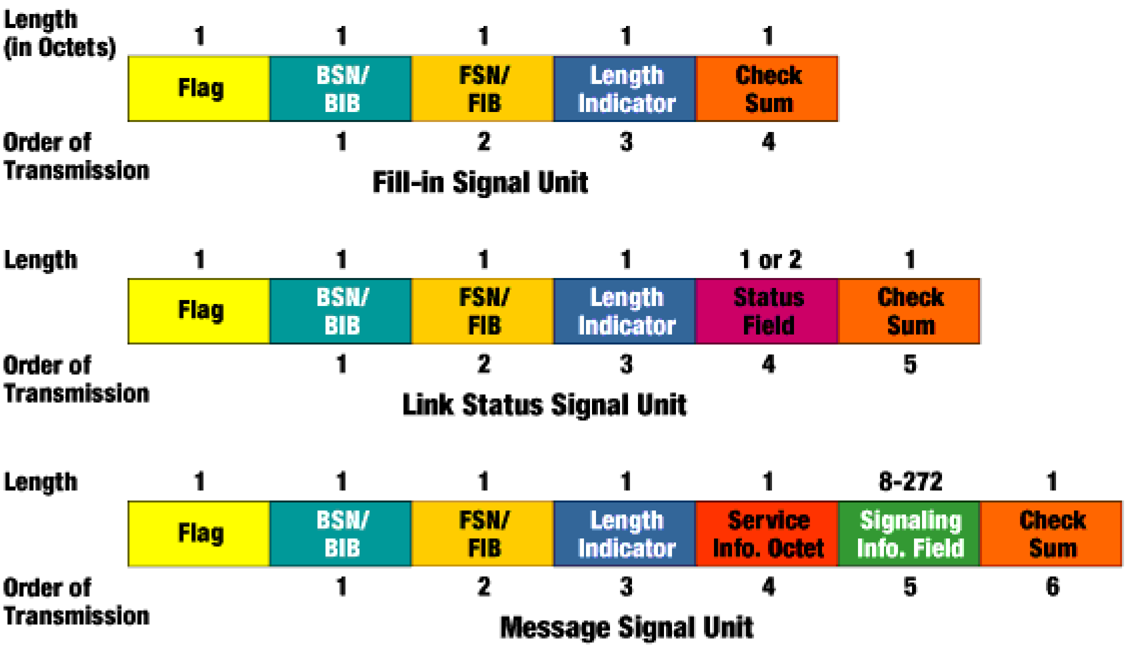
SUs are transmitted continuously in both directions on any link that is in service. A signaling point that does not have MSUs or LSSUs to send will send FISUs over the link. The FISUs perform the function suggested by their name; they fill up the signaling link until there is a need to send purposeful signaling. They also facilitate link transmission monitoring and the acknowledgment of other SUs.

All transmission on the signaling link is broken up into 8-bit bytes, referred to as octets. SUs on a link are delimited by a unique 8-bit pattern known as a flag. The flag is defined as the 8-bit pattern "01111110". Because of the possibility that data within an SU would contain this pattern, bit manipulation techniques are used to ensure that the pattern does not occur within the message as it is transmitted over the link. (The SU is reconstructed once it has been taken off the link, and any bit manipulation is reversed.) Thus, any occurrence of the flag on the link indicates the end of one SU and the beginning of another. While in theory two flags could be placed between SUs (one to mark the end of the current message and one to mark the start of the next message), in practice a single flag is used for both purposes.

Signal Unit Structure

SUs of each type follow a format unique to that type. A high-level view of those formats is shown in *Figure 8*.

Figure 8. Signaling Unit Formats



All three SU types have a set of common fields that are used by MTP Level 2. They are as follows:

Flag

Flags delimit SUs. A flag marks the end of one SU and the start of the next.

Checksum

The checksum is an 8-bit sum intended to verify that the SU has passed across the link error-free. The checksum is calculated from the transmitted message by the transmitting signaling point and inserted in the message. On receipt, it is recalculated by the receiving signaling point. If the calculated result differs from the received checksum, the received SU has been corrupted. A retransmission is requested.

Length Indicator

The length indicator indicates the number of octets between itself and the checksum. It serves both as a check on the integrity of the SU and as a means of discriminating between different types of SUs at level 2. As can be inferred from Figure 8, FISUs have a length indicator of 0; LSSUs have a length indicator of 1 or 2 (currently all LSSUs have a length indicator of 1), and MSUs have a length-indicator greater than 2. According to the protocol, only 6 of the 8 bits in the length indicator field are actually used to store this length; thus the largest value that can be accommodated in the length indicator is 63. For MSUs with more than 63 octets following the length indicator, the value of 63 is used.

BSN/BIB FSN/FIB

These octets hold the backwards sequence number (BSN), the backwards indicator bit (BIB), the forward sequence number (FSN), and the forward indicator bit (FIB). These fields are used to confirm receipt of SUs and to ensure that they are received in the order in which they were transmitted. They also are used to provide flow control. MSUs and LSSUs, when transmitted, are assigned a sequence number that is placed in the forward sequence number field of the outgoing SU. This SU is stored by the transmitting signaling point until it is acknowledged by the receiving signaling point.

Because the seven bits allocated to the forward sequence number can store 128 distinct values, it follows that a signaling point is restricted to sending 128 unacknowledged SUs before it must await an acknowledgment. By acknowledging an SU, the receiving node frees that SU's sequence number at the transmitting node, making it available for a new outgoing SU. Signaling points acknowledge receipt of SUs by placing the sequence number of the last correctly received and in-sequence SU in the backwards sequence number of every SU they transmit. In that way, they acknowledge all previously received SUs as well. The forward and backwards indicator bits are used to indicate sequencing or data-corruption errors and to request retransmission.

What are the Functions of the Different Signaling Units?

FISUs themselves have no information payload. Their purpose is to occupy the link at those times when there are no LSSUs or MSUs to send. Because they undergo error checking, FISUs facilitate the constant monitoring of link quality in the absence of signaling traffic. FISUs also can be used to acknowledge the receipt of messages using the backwards sequence number and backwards indicator bit.

LSSUs are used to communicate information about the signaling link between the nodes on either end of the link. This information is contained in the status field of the SU (see Figure 8). Because the two ends of a link are controlled by independent processors, there is a need to provide a means for them to communicate. LSSUs provide the means for performing this function. LSSUs are used primarily to signal the initiation of link alignment, the quality of received signaling traffic, and the status of the processors at either end of the link. Because they are sent only between the signaling points at either end of the link, LSSUs do not require any addressing information.

MSUs are the workhorses of the SS7 network. All signaling associated with call setup and tear down, database query and response, and SS7 network management takes place using MSUs. MSUs are the basic envelope within which all addressed signaling information is placed. As will be shown below, there are several different types of MSUs. All MSUs have certain fields in common. Other fields differ according to the type of message. The type of MSU is indicated in the service-information octet shown in Figure 8; the addressing and informational content of the MSU is contained in the signaling information field.

Message Signal Unit Structure

The functionality of the message signal unit lies in the actual content of the service information octet and the signaling information field (see *Figure 8*).

The service information octet is an 8-bit field (as might be inferred from its name) that contains three types of information as follows:

four bits are used to indicate the type of information contained in the signaling information field; they are referred to as the service indicator; the values most commonly used in American networks are outlined in *Table 1*

Table 1. Common Signaling Indicator Values

Value	Function
0	signaling network management
1	signaling network testing and maintenance
3	signaling connection control part (SCCP)
5	ISDN user part (ISUP)

two bits are used to indicate whether the message is intended (and coded) for use in a national or international network; they are generally coded with a value of 2, national network

the remaining 2 bits are used (in American networks) to identify a message priority, from 0 to 3, with 3 being the highest priority; message priorities do not control the order in which messages are transmitted; they are only used in cases of signaling network congestion; in that case, they indicate whether a message has sufficient priority to merit transmission during an instance of congestion or whether it can be discarded en route to a destination

The format of the contents of the signaling information field is determined by the service indicator. (Within user parts, there are further distinctions in message formats, but the service indicator provides the first piece of information necessary for routing or decoding the message.)

The first portion of the signaling information field is identical for all MSUs currently in use. It is referred to as the routing label. Simply stated, the routing label identifies the message originator, the intended destination of the message, and a field referred to as the signaling-link selection field which is used to distribute message traffic over the set of possible links and routes. The routing label consists of 7 octets that are outlined below in *Table 2* (in order of transmission).

Table 2. Routing Label

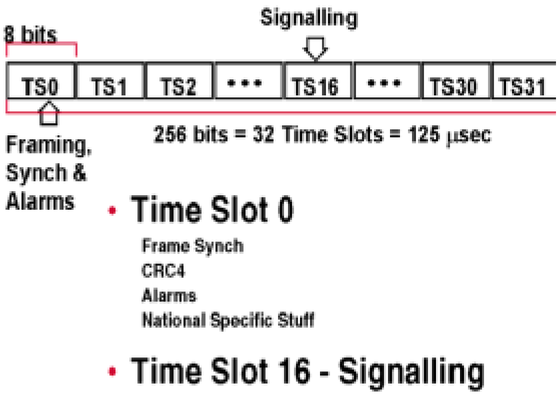
Octet Group	**Function**	**Number of Octets Involved**
destination point code (DPC)	contains the address of the node to which the message is being sent	3 octets
originating point code (OPC)	contains the address of message originator	3 octets
signaling link selection (SLS)	distributes load among redundant routes	1 octet

Point codes consist of the three-part identifier (network number, cluster number, and member number), which uniquely identifies a signaling point.

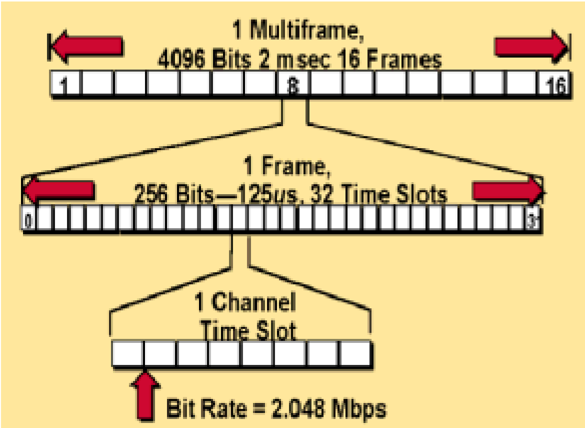
8.8. R2 Signaling

R2 signaling is a channel associated signaling (CAS) system developed in the 1960s that is still in use today in Europe, Latin America, Australia, and Asia. R2 signaling exists in several country versions or variants in an international version called Consultative Committee for International Telegraph and Telephone (CCITT-R2). The R2 signaling specifications are contained in International Telecommunication Union International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendations Q.400 through Q.490.

R2 signaling operates across E1 digital facilities. The E1 digital facilities carrier runs at 2.048 Mbps and has 32 time-slots. E1 time-slots are numbered TS0 to TS31, where TS1 through TS15 and TS17 through TS31 are used to carry voice which is encoded with pulse code modulation (PCM), or to carry 64 kbps data. The drawing below shows the 32 time-slots of an E1 frame.



An E1 carrier can use a multiframe structure within a Super Frame (SF) format or it can run in a non-multiframe mode without cyclic redundancy check (CRC). The SF format contains 16 consecutive frames numbered 0 to 15. Time-slot TS16 in frame 0 is used for SF alignment, and TS16 in the remaining frames (1-15) is used for CAS trunk signaling. TS16 uses 4 status bits designated as A, B, C, and D for signaling purposes. This multiframe structure is used for CRC, or error checking. This 16-frame multiframe structure (SF) allows a single 8-bit time slot to handle the line signaling for all 30 data channels. The following diagram illustrates the E1 SF format.



There are two elements to R2 signaling: Line Signaling (supervisory signals) and Interregister Signaling (call setup control signals). Most country variations in R2 signaling are with the Interregister Signaling configuration.

Line Signaling (Supervisory Signals)

You can use line signaling, which uses TS16 (bits A B C D), for supervisory purposes such as handshaking between two offices for call setup and termination. In the case of CCITT-R2 signaling, only bits A and B are used (bit C is set to 0 and bit D is set to 1). For two-way trunks, the supervision roles for forward and backward signaling vary on a call-by-call basis. The following table illustrates the R2 supervision signal, transition, and direction used on digital trunks.

An idle state is denoted when when A=1 and B=0.

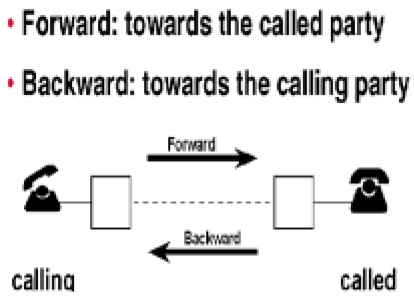
Direction	Signal Type	Transition
Forward	Seizure	A,B: 1,0 to 0,0
Forward	Clear-forward	A,B: 0,0 to 1,0
Backward	Seizure Acknowledgment (ACK)	A,B: 1,0 to 1,1
Backward	Answer	A,B: 1,1 to 0,1
Backward	Clear-back	A,B: 0,1 to 1,1
Backward	Release-guard	A,B: 0,1 to 1,0

Interregister Signaling (Call Setup Control Signals)

The concept of address signaling in R2 is slightly different from that used in other CAS systems. In R2 signaling, the exchanges are considered registers and the signaling between these exchanges is called inter-register signaling. Inter-register signaling uses forward and backward *in-band* multifrequency signals in each time-slot to transfer called and calling party numbers, as well as the calling party category.

Note: Some countries use two-out-of-six in-band dual tone multifrequency (DTMF) instead of forward and backward in-band multifrequency signals.

Multifrequency signals used during the Interregister Signaling are divided in forward signal groups (I and II), and backward signal groups (A and B). Interregister signaling starts after the 'Seize-ACK' of the line. The diagram and table below illustrate forward and backward signal information.



Forward Signal Groups	Backward Signal Groups
Group-I Signals Represent the called party number or dialed digits DNIS/ANI digits. I-1 to I-10 are digits 1 to 10. I-15 is the end of identification. **Group-II Signals** Represent the calling party category II-1 is subscriber without priority. II-2 to II-9 are subscriber with priority. II-11 to II-15 are spare for national use.	**Group-A Signals** Indicate if the signaling ended or if a particular forward signal is required. Used to acknowledge and convey signaling information A-1 is send next digit. A-3 is address-complete, changeover to reception of Group-B signals. A-4 is congestion. A-5 is send calling party's category. A-6 is address complete, charge, setup, speech conditions.

	Group-B Signals Sent by the terminating switch to acknowledge a forward signal, or to provide a call charging and called party information. Used to acknowledge Group-II forward signals. This is always preceded by an address-complete signal A-3. B-3 is subscriber line busy. B-4 is congestion. B-5 is unallocated number. B-6 is subscriber's line free charge.

The following inter-register group sequence rules are used to identify the signal's group:

The initial signal received by the incoming exchange is a Group I signal

Outgoing exchanges consider backward signals as Group A signals

Group A signals received by outgoing exchanges are used to identify whether the next signal is a Group B signal

Group B signals always indicate and end-of-signaling sequence

Most country-specific variations of R2 signaling are seen in the interregister signaling.