An HSS-PDL White Paper on Packet-SS7 by Sunil Mahajan

Abstract

The transition from circuit based voice telephony network to pure packet based voice telephony network requires huge investment. It also requires setting of a parallel packet voice telephony network components with definition of packet network signalling protocols and packet network bearer transport protocols and in addition, support for intelligent services on packet network.

This paper defines a Packet-SS7 network.

This paper tries to explore the idea of modifying existing SS7 network protocol and network architecture to support packet voice services on it. It can also provide an easy transition path from circuit voice network to pure packet voice network.

Voice Over Packet Network

Today's voice (and other media's as well) communication is moving away from circuit switch connection to packet connections for cost benefit, at the expense of voice quality on these networks. Typical network architecture of voice over packet network requires interworking function support at the edge of these two networks. These interworking functions for signalling and media interworking are supported by Voice over Packet Gateways. One of the popular Gateways (or technologies) among many is VoIP (voice over IP Network). During the transition phase, since the local connectivity to network will remain on PSTN ingress switches, the network architecture of a typical gateway cloud on PDN network will look as shown in following picture
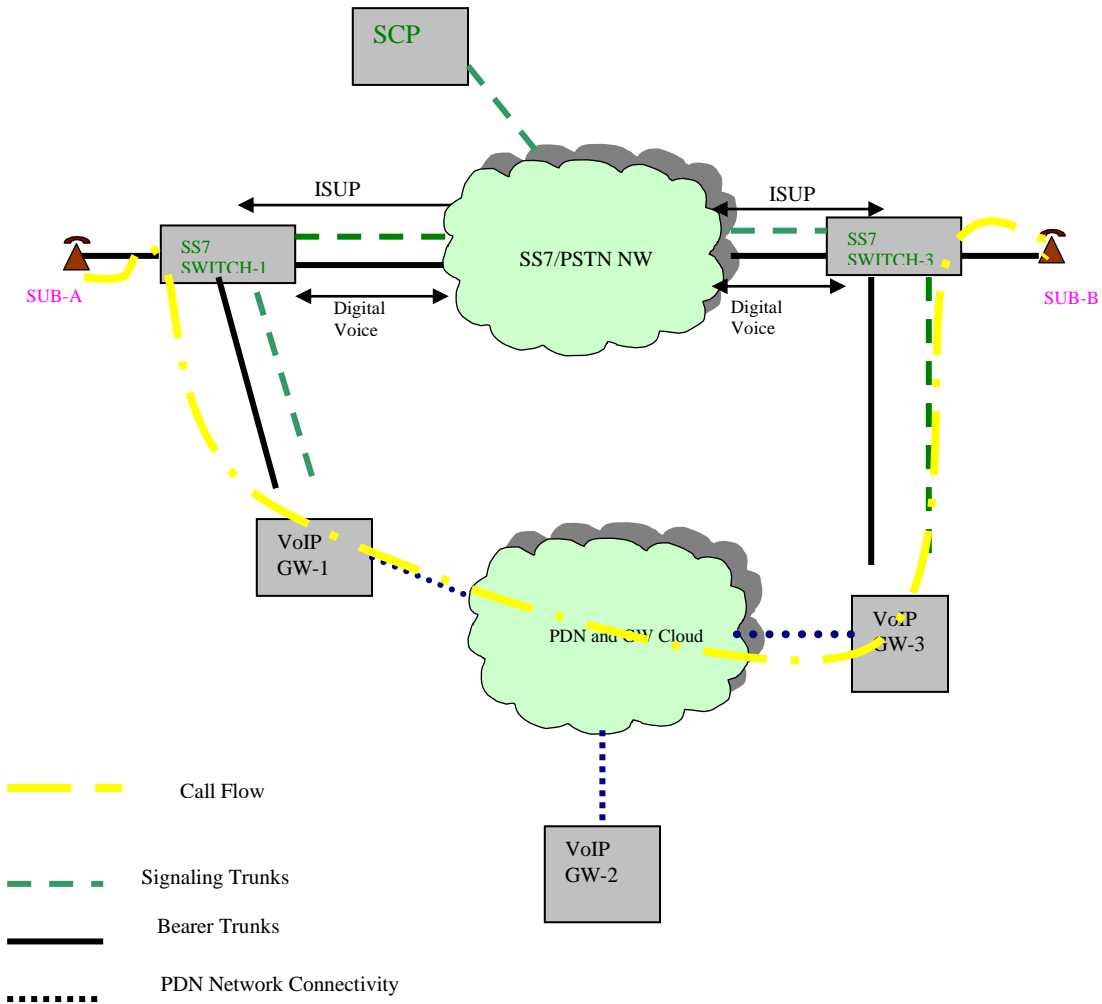
Figure 1 Voice over Packet network Architecture

Typical call routing with GW network

A typical call with above network requires offload of voice call from Ingress switch-1 to the gateway-1 and then the call gets routed to gateway-3 using packet signalling(e.g. H.323) and packet bearer protocols(e.g. RTP/RTCP) through PDN. Getway-3 routes the call back to Egress switch-3 and finally to subscriber.

Issues with GW network

There are few issues with above call senario

- Routing at GW-1 for call to SUB-B

- Address Translation at GW-1.

- Control of supplementary and intelligent services.

For all of the above issues there are solutions defined with these networks,
1. Routing and address translation decision is taken either at gateway-1, if it maintains the network routing database and address translation tables.
2. Routing and address translation decision is taken by an entity called Gatekeeper (GK), which holds the routing database and translation database.
3. Routing and address translation is done through GW-1 or through GK cloud by making a query to AIN network (over SS7 network) entity, called SCP (service control point).
4. Control of supplementary services and intelligent services is through GK or through SCP (over SS7 network).
5. Signalling may be through PDN is directly between GW-1 and GW-2.
6. Signalling may be controlled by GK cloud.

Above solution requires following entities in the Gateway network

- Gatekeeper Network

- Connectivity to AIN network over SS7 or a parallel AIN network support

Moreover this kind of network requires building parallel telephony components.
For example

- AIN components on PDN.

- signalling paths or routes on PDN

- signalling protocols on packet network

- subscriber database for packet subscribers etc.

However this can be avoided, if we use the existing SS7 network or PSTN network on SS7, for both packet signalling and packet bearer communication. We will be discussing the architecture of such network after we discuss in brief network architecture SS7 network.

Today's SS7 Circuit Switched Voice Network
The SS7 voice network is depicted in following picture
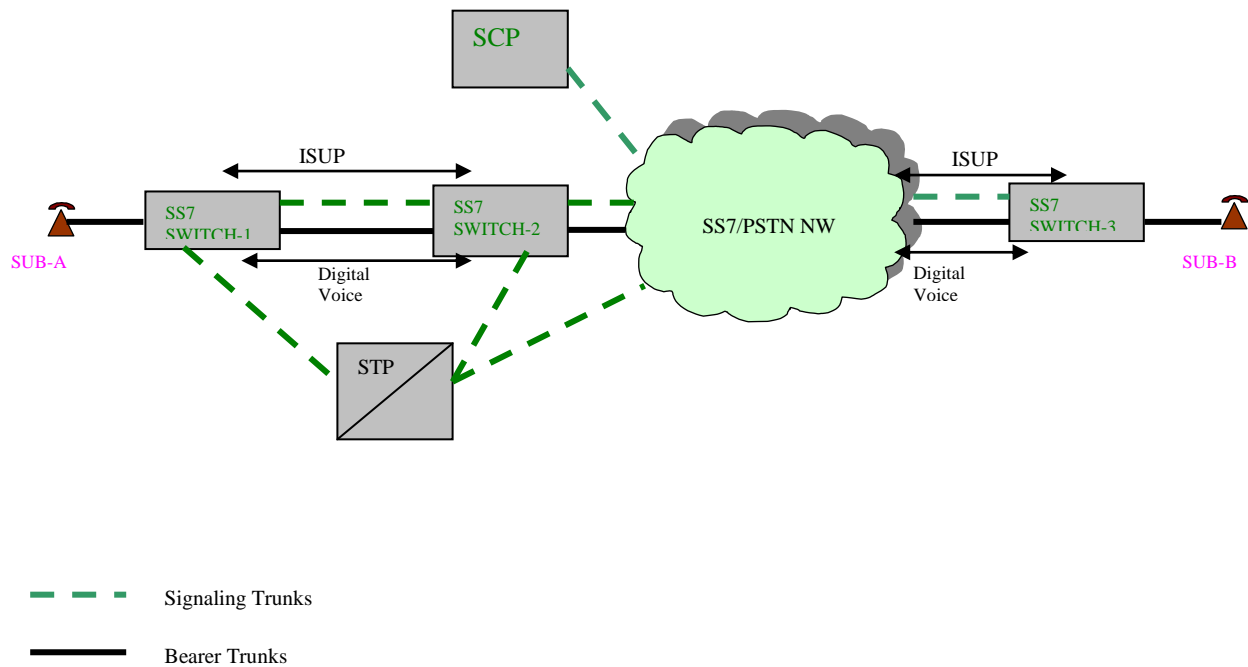
Figure 2 SS7 Circuit Switched Voice Network

A typical call establishment from SUB-A to SUB-B on SS7 network, requires following steps

- Switch-1 detects the call attempt by SUB-A

- Switch-1 decides the call route to reach Switch-3 for SUB-B.

- Route to Switch-3 may be decided locally by call control or by remote database query to SCP over SS7 network.

- Deciding route involves selecting next switch on the route and selecting a circuit (CIC) based on the bearer channel requirement.

- Switch-1 selects a voice trunk (or voice circuit) to Switch-2.

- Switch-1 sends ISUP signalling information to Switch-2.

- Switch-1 switches the voice path to Switch-2.

- On receiving ISUP signalling information the routing, signalling and switching procedure is repeated and this way call reaches finally to Switch-3.

- Switch-3 does local (access side) call signalling to SUB-B and switches the path to subscriber if it is free.

- Switch-3 sends signalling information back to Switch-1 through Switch-2.

- Switch-1 connects the subscriber to voice trunk.

- Call is established.

In this call establishment following are the points to be noted

- Routing database is distributed through out the network.

- Each node decides the next node in the path.

- Infrastructure is in place for circuit establishment and signalling path.

- Path and protocols to intelligent network entities (SCP etc.) are established.

Using SS7 Network for Packet communication

There are three alternatives for packet communication using SS7 network infra-structure

1. Use existing SS7 infrastrucure(SS7 switches and other nodes) for routing, supplementary services, intelligent network functions. Use SS7 signalling trunks for signalling and use SS7 bearer trunks for packet communication. *Call this PSA-1 (Packet SS7 Architecture-1).*

2. Use existing SS7 infrastrucure(SS7 switches and other nodes) for routing, supplementary services, intelligent network functions. Use SS7 signalling trunks for signalling. Provide PDN network connectivity to SS7 nodes for bearer communication. *Call this PSA-2 (Packet SS7 Architecture-2).*

3. Mix of above two architectures, with SS7 nodes supporting both kind of connectivities. *Call this PSA-3 (Packet SS7 Architecture-3).*

We will discuss each of these architectures in details in subsequent sections.

PSA-1

This architecture supports the services, signalling and packet voice or bearer communication on the existing SS7 infrastructure. The modifications required for this support is in following modules

- Call Control Function

- Enhancement of ISUP protocol

- Support for packet protocol on circuit connection.

- Firmware to support media transcoding.

The architecture first will be discussed in view of interaction between two adjacent SS7 switches and then will be further extended to the network.
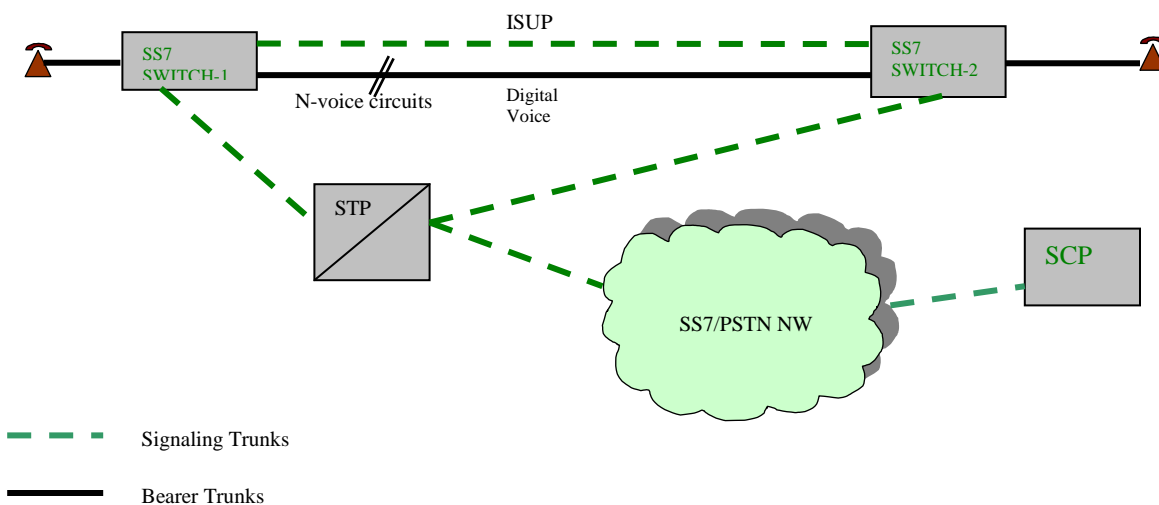

Figure 3 SS7 Voice Network

The above configuration contains two switches connected by signalling trunks directly connecting these two switches and also through an alternate-signalling path. N-voice circuits connect these two switches, which directly maps to N-cics between these two switches. However these N-circuits may have few digital circuits on E1/T1s, few on satellite or may be few on low speed links etc. So a typical call establishment between these two switches requires selection of a CIC based on the bearer channel requirement. Now consider another configuration
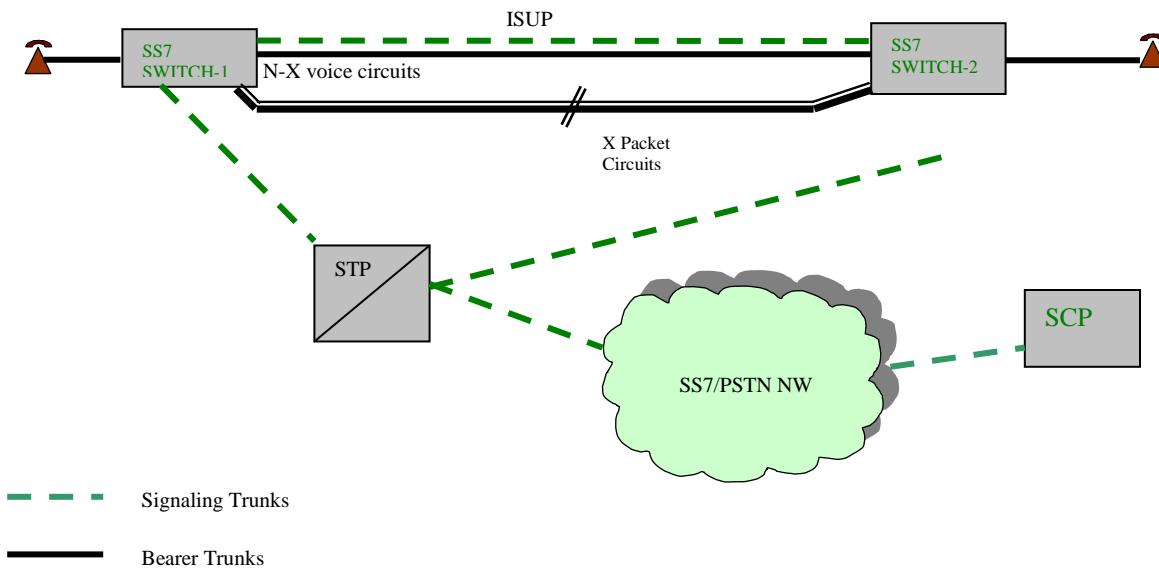
Figure 4 Packet SS7 Network Architecture-1

This configuration has divided N-voice circuits to two groups, first group contains N-X voice circuits and the other group contains X packet circiuts.

Packet Circuit: Packet circuits are the same digital voice circuits between two switches but supports packet communication. To support packet communication on these digital circuits, the two switches connecting these circuits run HDLC protocol on these circuits. The HDLC protocol however is not run on individual circuit but on a group of circuit bundled together to make a packet-circuit group. This grouping depends on the physical grouping of these circuits, for example all circuits on an E1 trunk makes one packet group.

Configuration of packet circuits

The two switches connecting these packet circuits needs to assign cics for them. The number of CICs assigned can be more than the number of circuits in these packet groups. A typical configuration may assign cics in the range of 8-10 times the number of physical circuits. This assignment is based on the assumption that packet circuits can support more calls than their physical number, for following reasons

- Compressed voice on these circuits

- Silence suppression results in less bandwidth requirements.

• Replaying last received compensates lost voice packets.

Voice transport on packet circuits

Transport of voice on these packet circuit uses the mechanism recommended for H.323 networks. The transport uses Real-Time Transport Protocol (RTP) and Real-Time Transport Control Protocol (RTCP). As the requirement from these protocol requires an un-reliable lower layer transport channel, which in most of the network is provided by UDP/IP protocol in the network. The confguration suggested above does not require any routing of packets at lower layer, since the connectivity is point to point and the identification of call in RTP session will be done by small ss7-packet header. So a typical RTP packet will look like

HDLC HDR	SS7-PAC-HDR	RTP HDR	RTP PAYLOAD	HDLC TRL
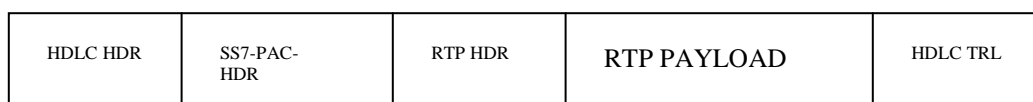
HDLC and RTP HDR are the standard headers, but SS7 header will look like as shown in following picture (it is like a routing label in SS7 message with CIC added).

OPC	DPC	CIC
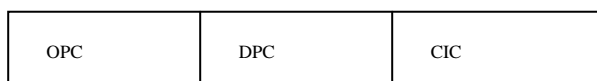
Call Establishment on packet circuit

When two exchanges comes up they block all the packet circuits till HDLC communication is established on them. After HDLC channel is established, the circuits are unblocked and are available to call control for allocation. At any time if the HDLC communication goes away or channel is diconnected the affected CICs are blocked by both the exchanges.

Support for H.245 signalling

Before RTP channel communication a typical packet network signalling requires capabilities exchange between two nodes to establish the characterstics of the voice communication supported, for example the type of voice coding used, jitter buffer parameters etc. This can be

achieved by either *assuming a static configuration*[1] of these capabilities at both the switches during the CICs initialisation or exchange of this information during call establishment, i.e. the forward direction capabilities parameters are send with IAM message by introducing another parameter in IAM message

PARAMETER NAME= ISUP_FORW_CAPABILITY_SET
PARAMETER_LEN = X Bytes
PARAMETER CONTENTS as specified in H.245 (ASN encoded)

Moreover to indicate that capability exchange is required before call establishment, Nature_Of_Connection_Indicator parameter in IAM is modified as

8	7	6	5	4	3	2	1
H	G	F	E	D	C	B	A

Bit F : Capability Exchange Required

   0   Capability Exchange Not Required

   1   Capability Exchange Required

If capability exchange is required the originating exchange will wait for backward message which is defined as

Message Type: Backward Capability Exchange (BCE)

Parameters : Mandatory Variable Length: ISUP_BACW_CAPABILITY_SET

           Length: X Bytes

           Contents: ASN formatted capability set

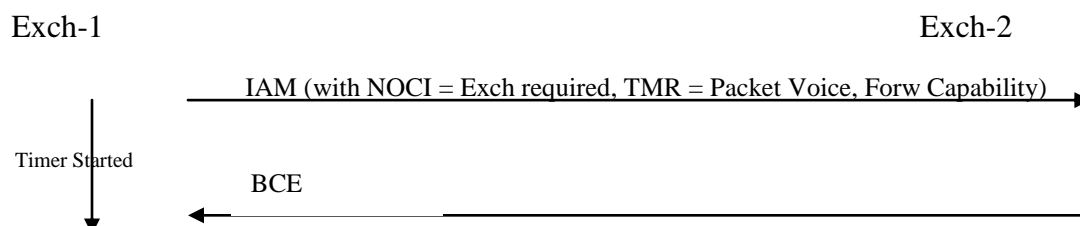
[1] *This is prefered.*

A typical call establishment

A typical call establishment between two exchange will require following steps

1) Exch-1 based on the subscriber profile and bearer channel requirements from the access side will decide whether to use packet circuits or voice circuits.

2) If there is no packet circuit available then Exch-1 can select a voice circuit for the call.

3) Assume Exch-1 selects packet circuit.

4) Exch-1 selects a packet CIC.

5) Exch-1 based on the cic configuration decides to exchange the capability set for the call.

6) Exch-1 generates an IAM with following parameter parameters

TMR (Transmission Medium Requirements): Voice Packet Circuit

NOCI (Nature of Connection Indicator): Capability Exchange Required

ISUP_FORW_CAP: Forward capability set

Exch-1   Exch-2

IAM (with NOCI = Exch required, TMR = Packet Voice, Forw Capability)

Timer Started

BCE

7) Exch-2 on deciding capability set generated message BCE back to Exch-1

8) Both the exchanges set the RTP and RTCP protocol for this cic.

9) After these message exchange, the call establishment is like a normal ISUP call establishment.

10) After the complete call path is established RTP packets are exchanged between these two exchanges for voice or bearer communication.

11) RTCP is used to control the RTP sessions. It can also be used to control the quality of service for RTP sessions. If RTCP reports more packet loss, congestion and delays, then either exchange can block few of the CICs to reduce further traffic between them.

Extending PSA-1 to network

PSA-1 architecture discussed above involves only two exchanges. The following configuration is assumed in the network for packet communication across network.
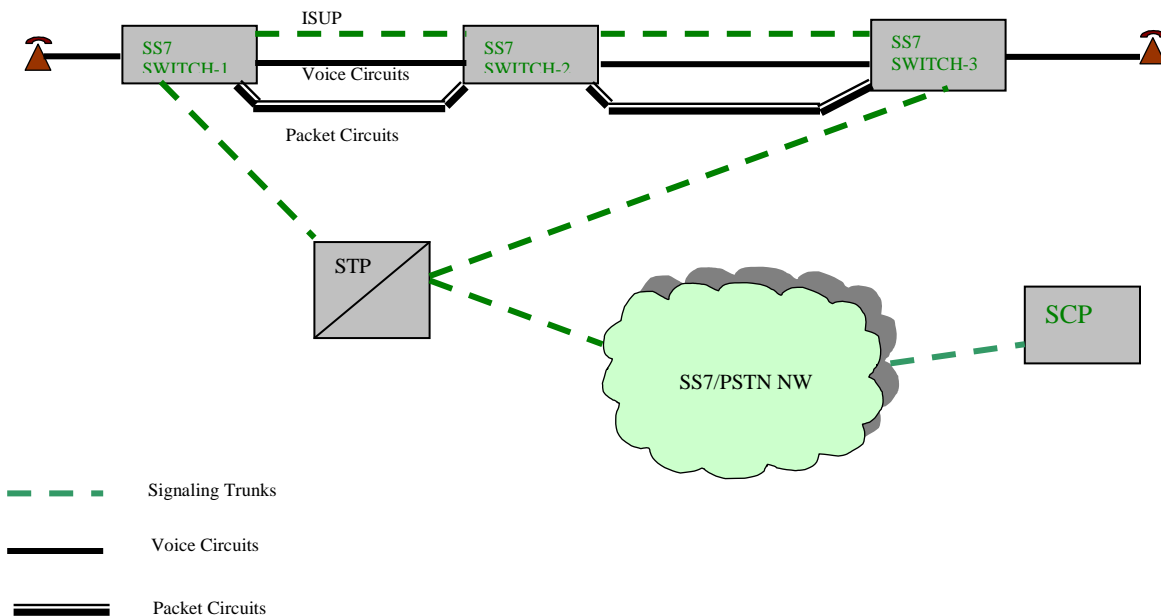

Figure 5 Extension to PSA-1

The difference between this configuration and previous configuration is that , a call establishment in this case requires signalling information exchange between Exch-1, Exch-2 and Exch-3. Further there will be two RTP sessions involved, one between Exch-1 and Exch-2 and other between Exch-2 and Exch-3. In addition to all the fucntions as required for previous configuration, this configuration requires RTP relay function at Exch-2.

RTP-Relay Function[2]

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[2] *RTP Relay function can be implemented in firmware for fast switching*

RTP relay function requires terminating RTP protocol at one end and relaying or switching RTP packets to other side. As discussed earlier RTP exchange between two Exchanges carries an SS7-PAC-HDR which contains routing label information. So at the time of RTP session establishment on both sides RTP relay function needs to maintain the routing label mapping for both sides and during data transfer phase can use this map table to switch RTP packets. RTP relaying requires changing SS7-PAC-HDR and optionally requires changing RTP header.

Most of the media processing (echo cancellation etc.) related functions can be done at the originating exhange or at terminatig exchange, but they may optionally be done at realy nodes as well. However few of these functions may be done at all nodes, for example jitter management, packet loss compensation.

PSA-2

This network architecture uses existing SS7 infrastrucure(SS7 switches and other nodes) for routing, supplementary services, intelligent network functions. It uses SS7 signalling trunks for signalling. However it uses PDN network connectivity to SS7 nodes for bearer packet communication.
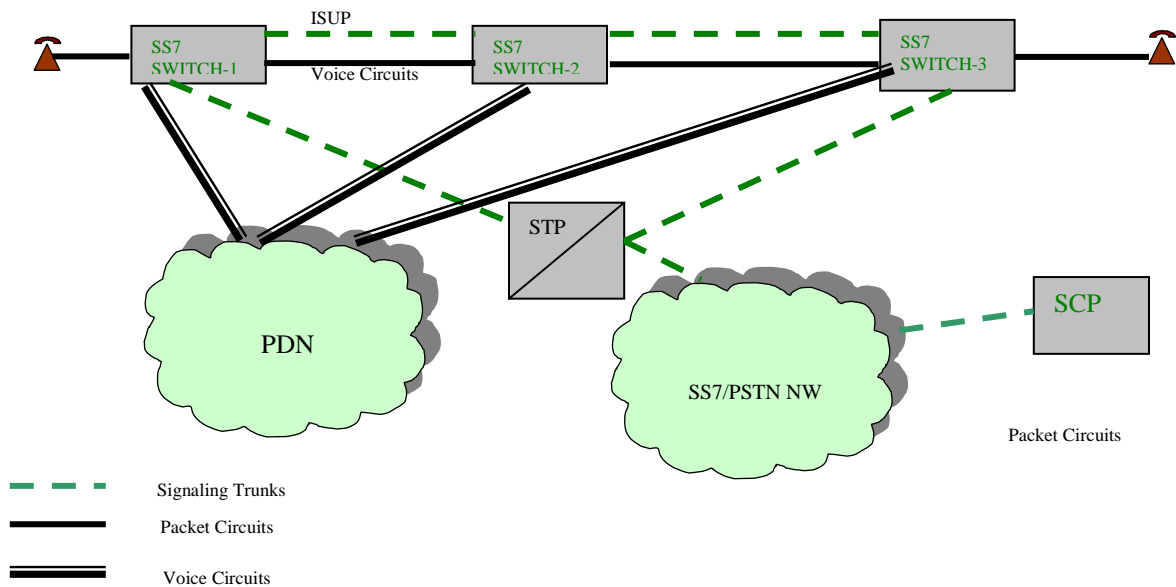

Figure 6 Packet SS7 Architecture-2

In first of these configuration each of the switch is connected PDN network for packet voice connectivity, which carries the complete RTP/RTCP session on UDP/IP network. The CIC configuration for this connectivity may use a different OPC/DPC combination, other than

OPC/DPC pairs for voice circuits or it may use the same OPC/DPC pair. The number of CICs configured may depend on the kind of network connectivity and bandwidth available.

This configuration supports a packet structure similar to the one described earlier.

IP	UDP	SS7-PAC-HDR	RTP HDR	RTP Payload

This RTP communication however does not require establishment of HDLC channel between two exchanges. RTCP in this configuration can still be used for congestion control to block circuits in case of high packet loss and congestion in the network. This configuration still requires all the support as described in PSA-1 for ISUP (TMR modification, NOCI modification, BEC message support, Forward and Backward capability parameters).
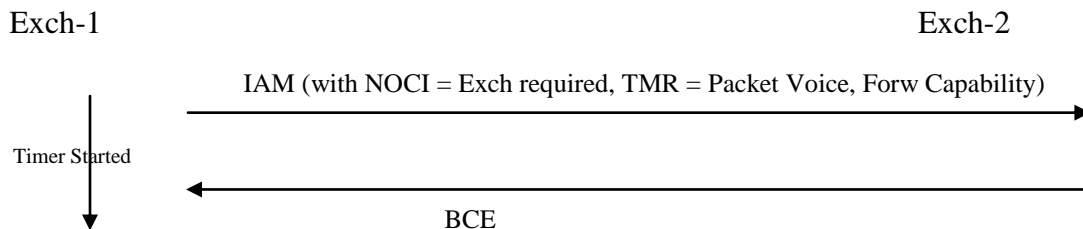
A typical call establishment

A typical call establishment between two exchange will require following steps

1) Exch-1 based on the subscriber profile and bearer channel requirements from the access side will decide whether to use packet circuits or voice circuits.

2) If there is no packet circuit available then Exch-1 can select a voice circuit for the call.

3) Assume Exch-1 selects packet circuit.

4) Exch-1 selects a packet CIC.

5) Exch-1 based on the cic configuration decides to exchange the capability set for the call.

6) Exch-1 generates an IAM with following parameter parameters

   TMR (Transmission Medium Requirements): Voice Packet Circuit

   NOCI (Nature of Connection Indicator): Capability Exchange Required

   ISUP_FORW_CAP: Forward capability set

Exch-1   Exch-2

IAM (with NOCI = Exch required, TMR = Packet Voice, Forw Capability)

Timer Started

BCE

7) Exch-2 on deciding capability set generated message BCE back to Exch-1. Capability exchange between two exchange equires exchange of UDP ports for RTP/RTCP sessions.

8) Both the exchanges set the RTP and RTCP protocol for this cic.

9) After these message exchange, the call establishment is like a normal ISUP call establishment.

10) After the complete call path is established RTP packets are exchanged between these two exchanges for voice or bearer communication.

11) RTCP is used to control the RTP sessions. It can also be used to control the quality of service for RTP sessions. If RTCP reports more packet loss, congestion and delays, then either Exch-2 can block few of the CICs to reduce further traffic between them.

Modification to above configuration

Above configuration can be modified to support long PDN connectivity as shown in following picture.
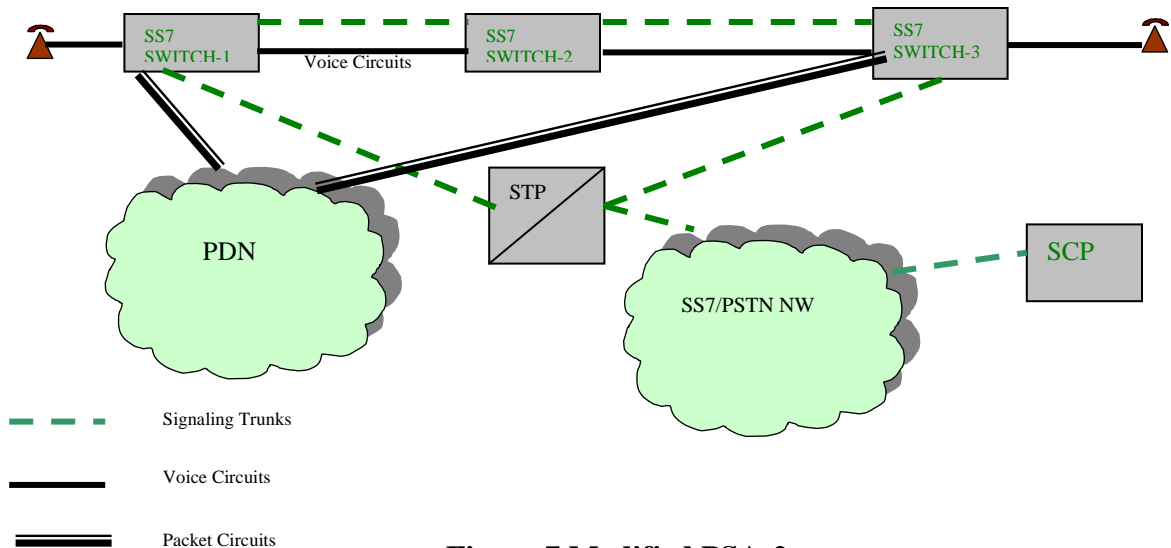

Figure 7 Modified PSA-2

In this configuration Switch-1 and Switch-3 are adjacent switches for packet CICs and can have direct signalling and packet bearer establishment. Rest of the procedures for call establishment and bearer communication are similar to as defined with previous configuration.

PSA-3

Architecture PSA-3 is combination of above two architecture with signalling and packet bearer communication supported as required for each of them.
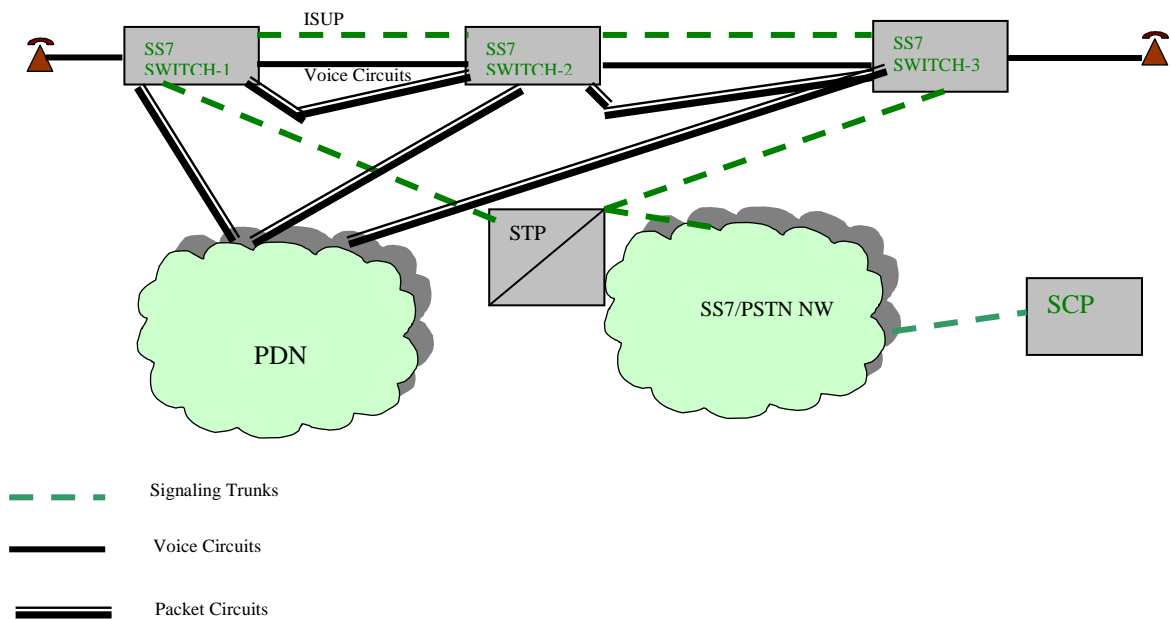

Figure 8 Packet SS7 Architecture-3

This configuration supports packet circuit connections between two switches directly and also through PDN.

Conclusion

Packet-SS7 architecture may provide a transition solution from circuit network telephony to pure packet network telephony. With the number of circuits X (packet voice circuits) increasing with each telephony switch to finally N (total voice circuits), ciruit switches can be converted to packet switches. Moreover the architecture PSA-3 or PSA-2 may provide a smooth transition from circuit telephony to packet telephony transition with voice switches replaced by packet switches supporting packet signalling protocols and packet bearer protocols.

References

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ITU-T Q Series Recommendation Q.122x, Intelligent Networks
Interface Recommendation for Intelligent Network CS-2

ITU-T Q Series Recommendation Q.76x, SS7
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Visual Telephone Systems and Equipment for Local Area Networks which provides a Non-Guaranteed Quality of Service

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Call Signalling Protocols and Media Stream Packetization for Packet Based Multimedia Communications Systems

ITU-T H Series Recommendation H.245,
Line Transmission of Non-Telephone Signals

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Acknowledgements