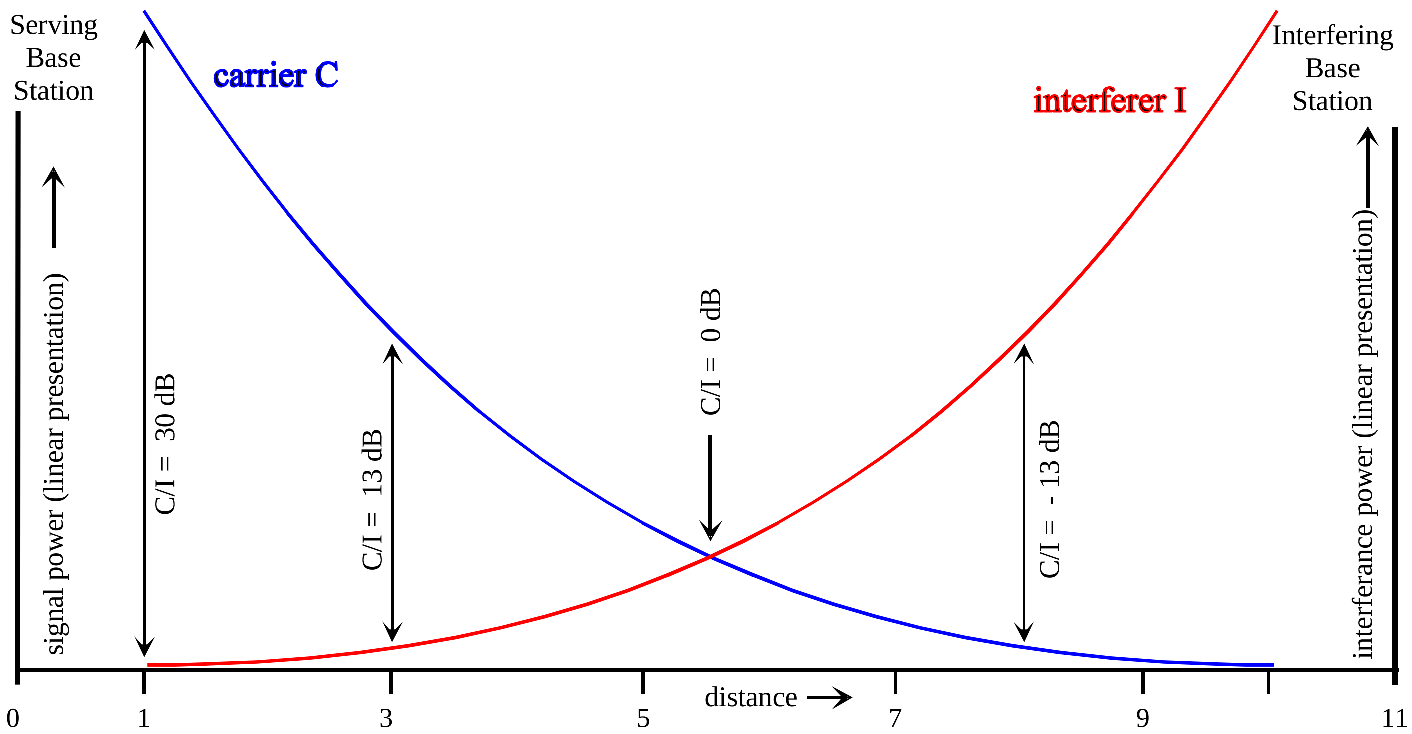
Mobile Radio

Special Subjects

1 Radio Network Planning

1.1 Carrier to Interference Ration

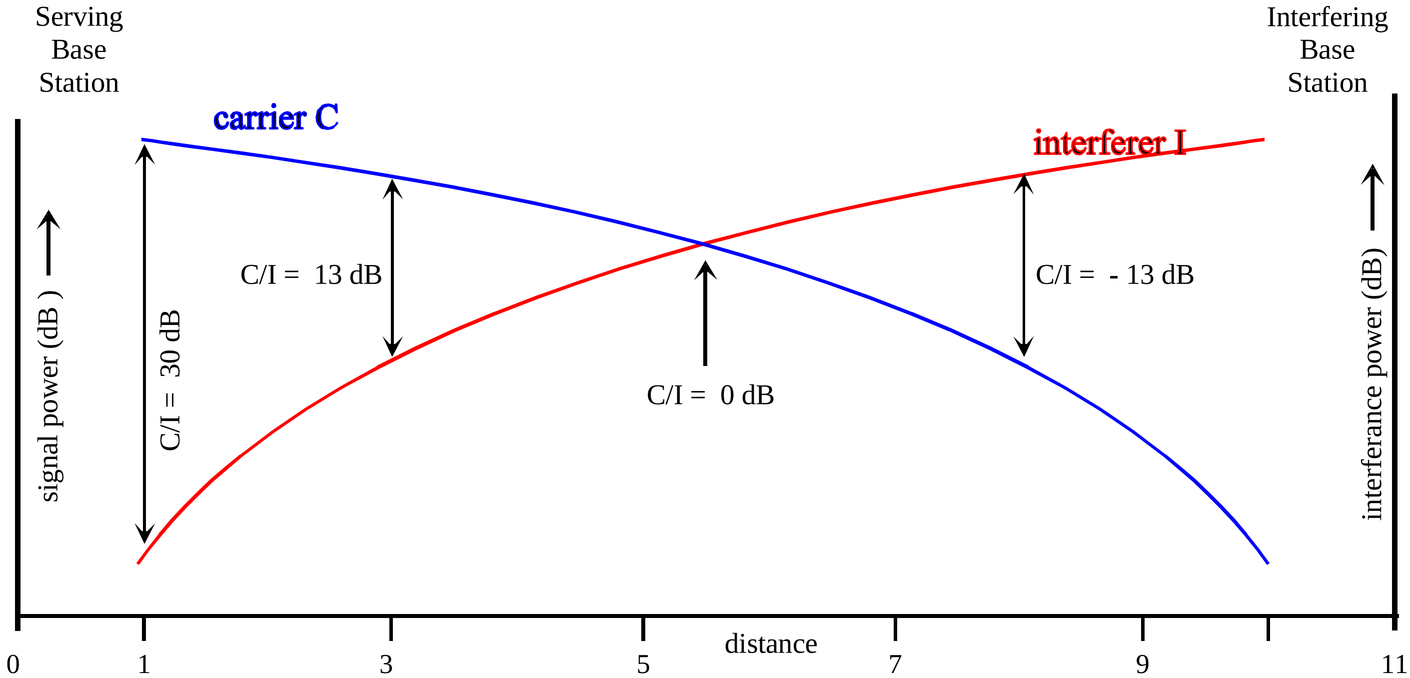
2 base stations using the same frequency and having the same transmit power



Carrier to Interference Ration

2 base stations using the same frequency and having the same transmit power

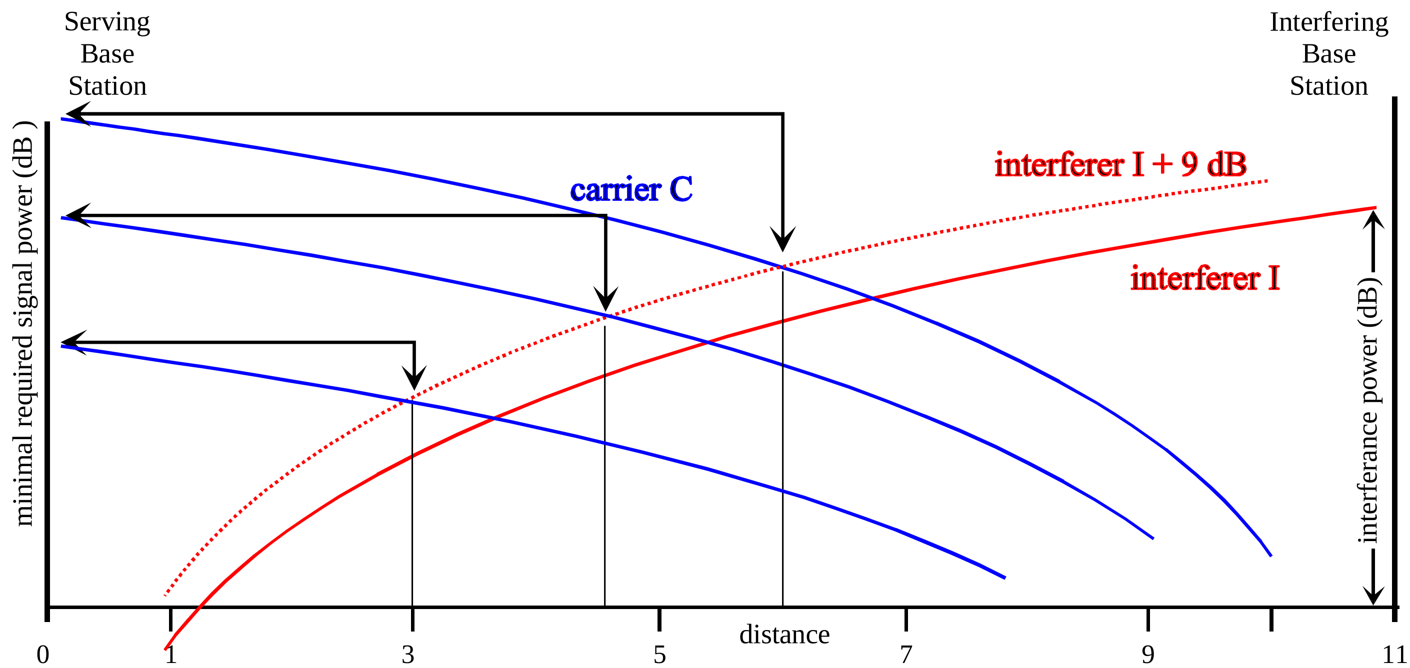
logarithmic power presentation [dBm]



Minimal Required Signal Power

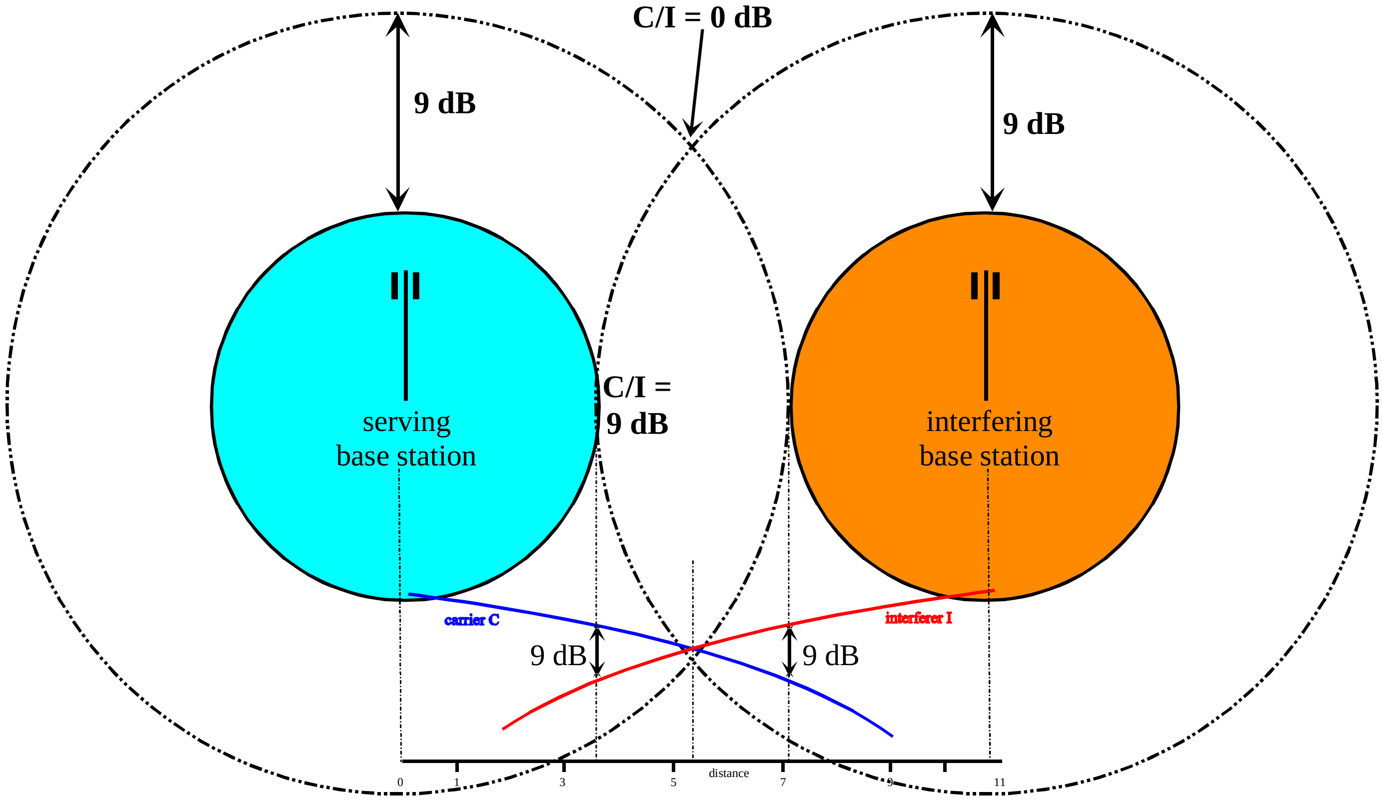
logarithmic power presentation [dBm]

assumption: C/I $\geq$ 9 dB



Co-channel Interference

two base stations having the same transmission power and frequency



1.2 Cell Planning

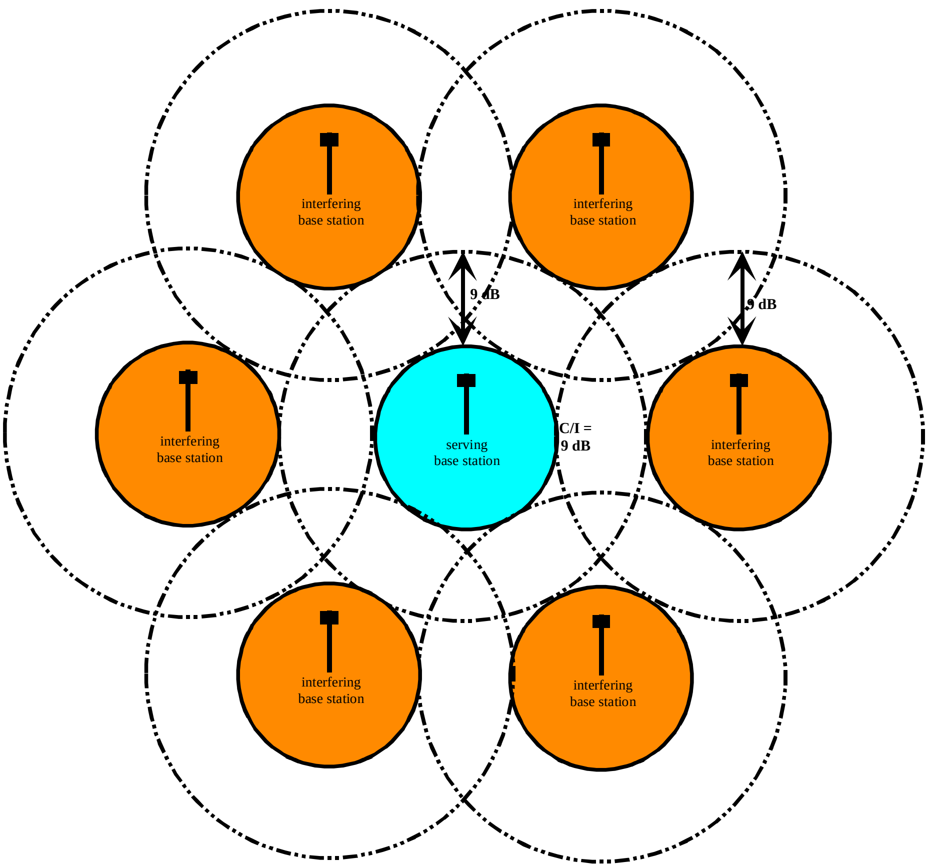
a pattern has to be found,

witch fits

the C/I requirements between all cells using the same frequency

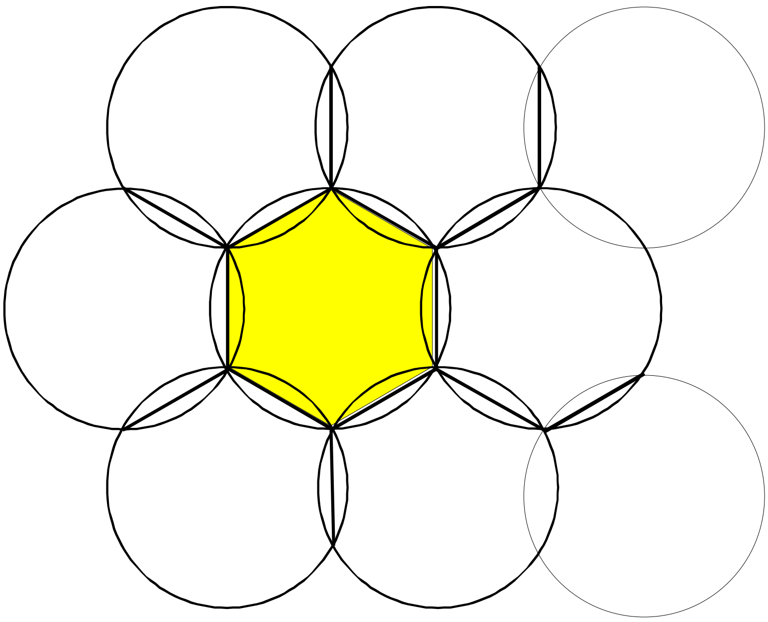
witch fills

the space between the cells with cells of other frequencies regarding the C/I requirements set above.

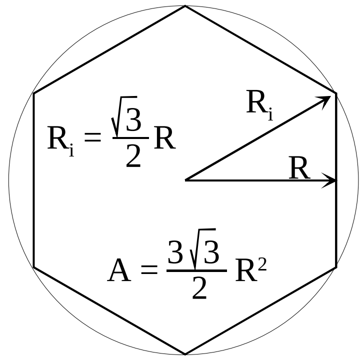


Circles are turning into Hexagons

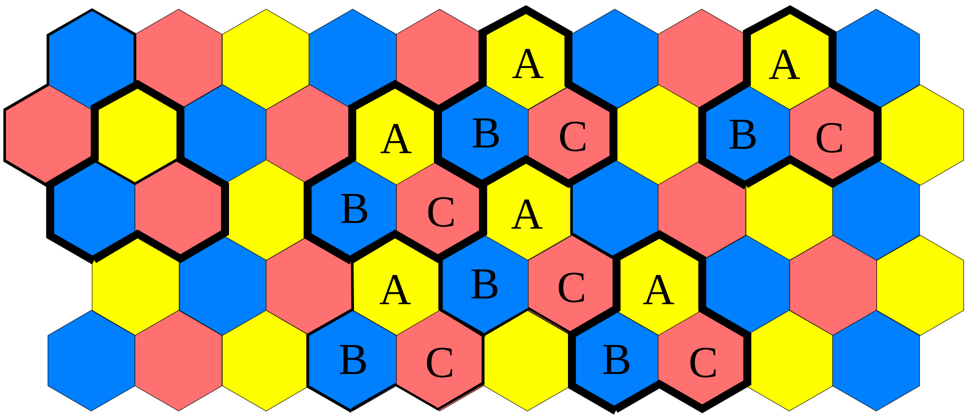
the radio cells have to cover the area completely



simplified modell of radio cells
with reglar hexagons

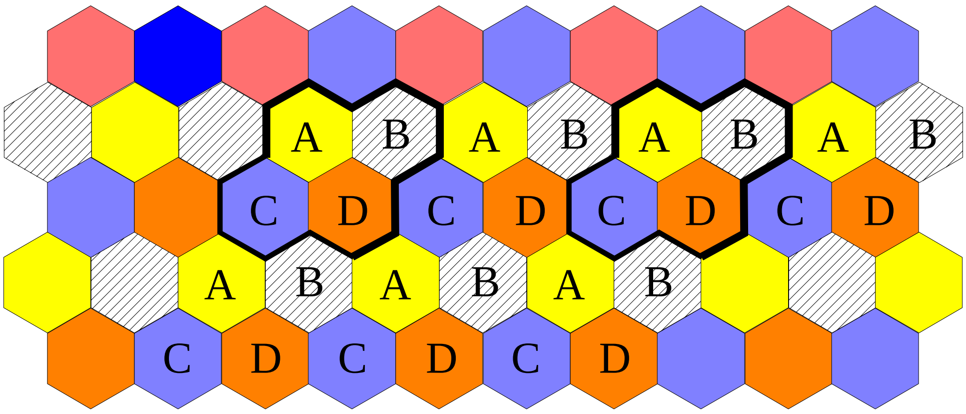


In practice, radio cells show irregular forms



Cluster of 3

Cluster of 4

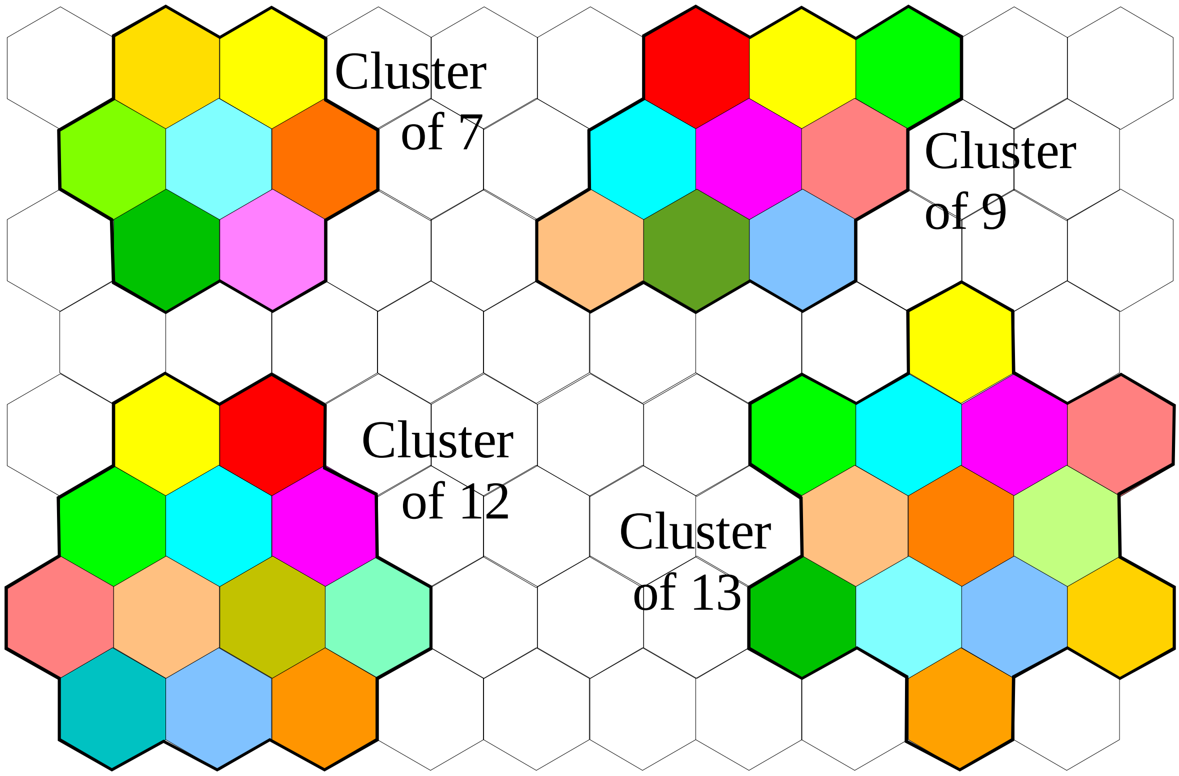


1.3 Cluster

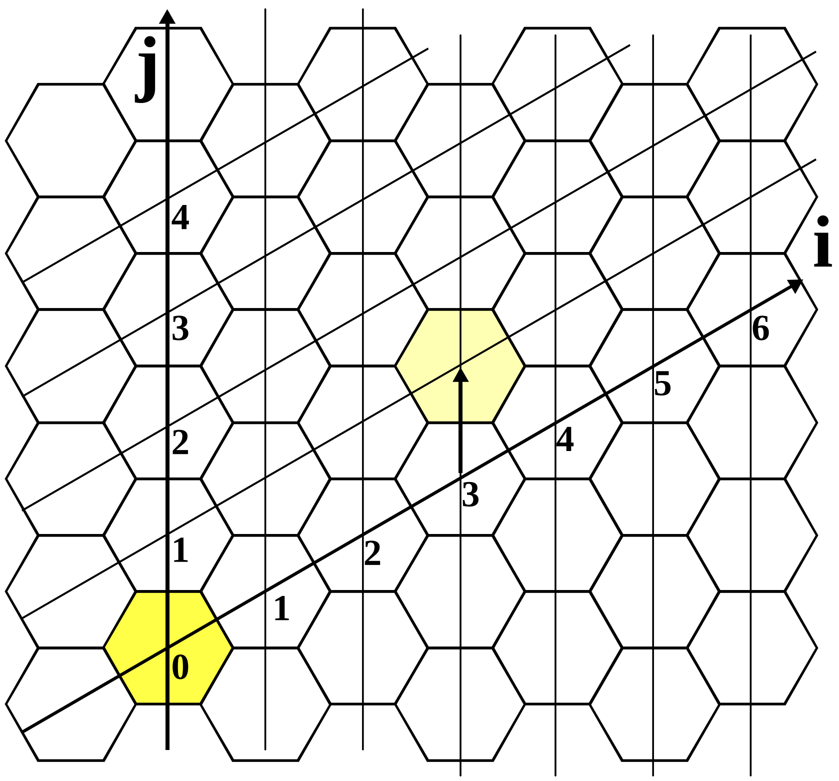
a group of cells. When this group is repeated regularly, an area can be covered completely.

the distance between corresponding cells is always the same.

Cluster Size 7, 9, 12, 13



Cluster Size in a 60°-(i,j)- Coordinate System



Cluster Size $N = i^2 + i\,j + j^2$

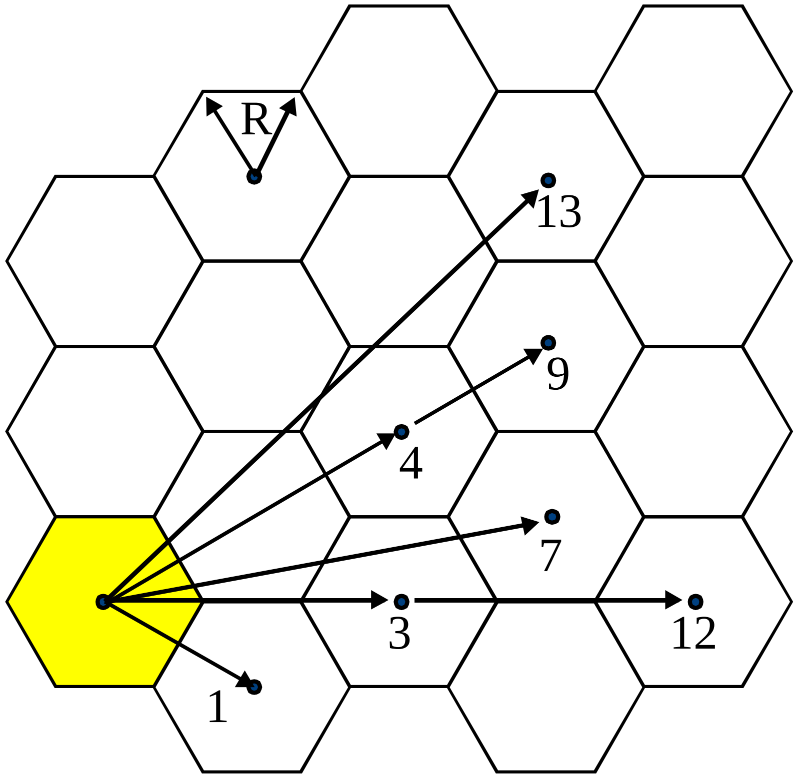
i, j Integers

Example: i = 3; j = 1

$3^2 + 3*1 + 1^2 = 9 + 3 + 1 = 13$

➔ the cells marked yellow in the figure fit into a cluster with 13 cells.

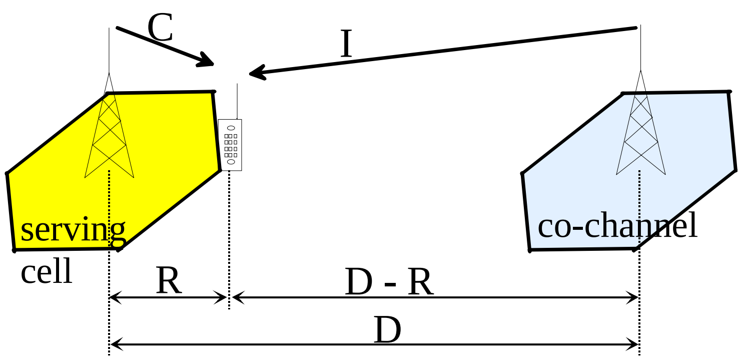
Examples of Clusters



q = Distance / Cell Radius R

Cluster Size N	q = D/R
1	1,73
3	3
4	3,46
7	4,58
9	5,2
12	6
13	6,24

Carrier to Interference Ratio C/I



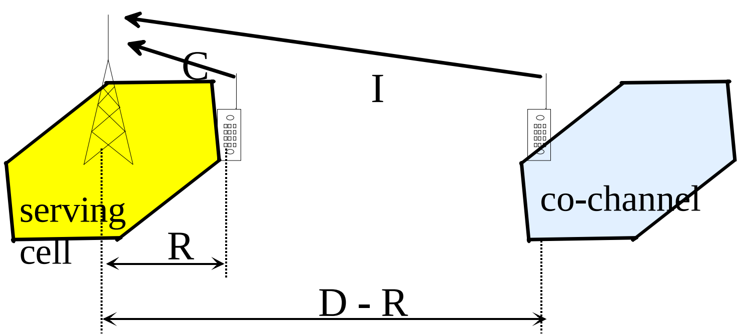
Downlink Interference BS ➔ MS

$$\frac{C}{I} = \frac{R^{-4}}{(D-R)^{-4}}$$

only 1 interferer, in reality up to 6 interferers

Uplink Interference MS ➔ BS

$$\frac{C}{I} = \frac{R^{-4}}{(D-R)^{-4}}$$



Carrier to Interference Ratio C/I

C/I - Estimation:

Pessimistic: $\frac{C}{I} = \frac{R^{-4}}{6\,(D-R)^{-4}} = \frac{1}{6\,(q-1)^{-4}}$

Normal: $\frac{C}{I} = \frac{R^{-4}}{6\,D^{-4}} = \frac{1}{6\,q^{-4}} = 1{,}5\ N^2$

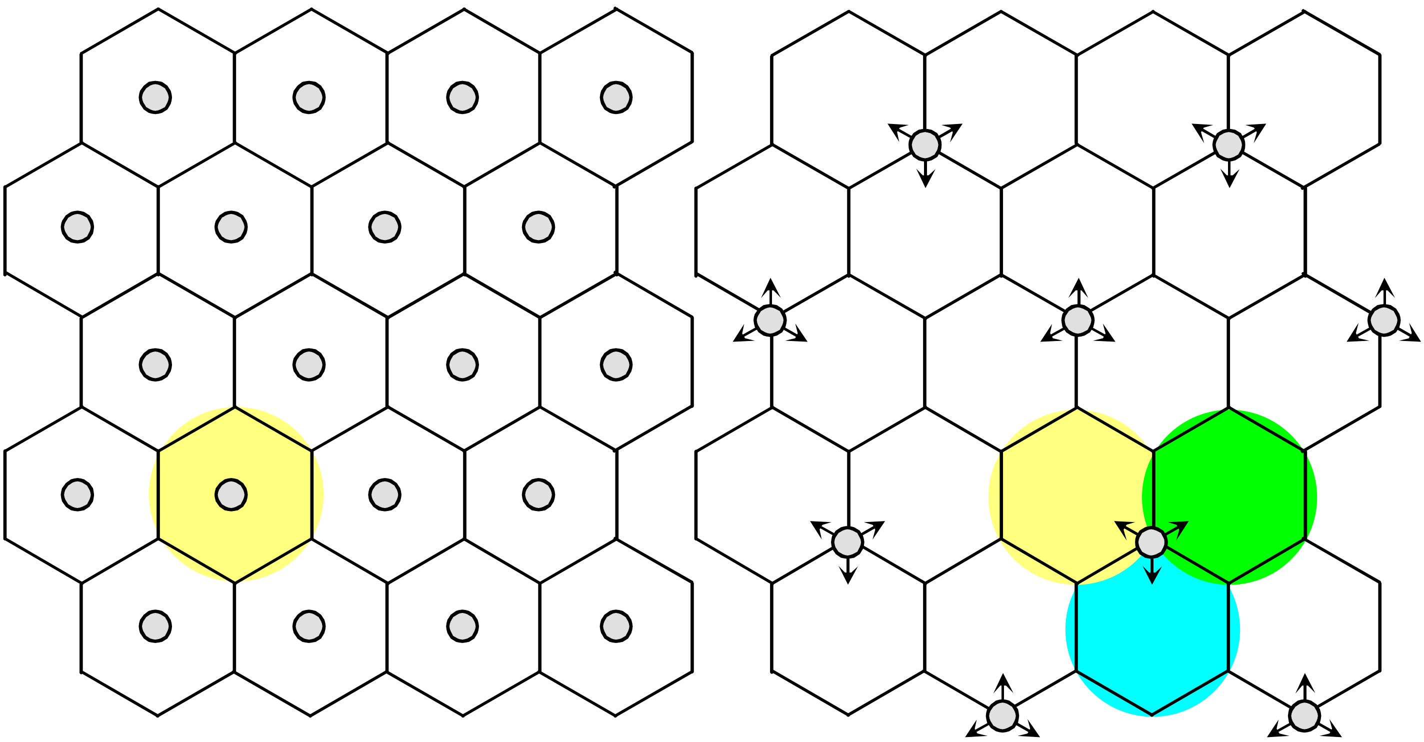
C Carrier,
I Interferer,
N Cluster Size
D Frequency Reuse Distance
R outer Cell Radius
q $= D/R = \sqrt{3\,N}$

N	q	C/I Pessimist.	C/I Normal
1	1,73	-13,20	1,76
3	3,00	4,26	11,30
4	3,46	7,88	13,80
7	4,58	14,39	18,66
9	5,20	17,13	20,85
12	6,00	20,18	23,34
13	6,24	21,01	24,04
16	6,93	23,14	25,84
19	7,55	24,87	27,34
21	7,94	25,87	28,21
27	9,00	28,34	30,39

Cell Splitting

1 Cell per BS Site

3 Cells per BS Site



homogene Zellstruktur und Realität

Vorteil der geometrischen Methode:

- mathematische Behandlung
- gutes Verständnis in die Prinzipien
- gut geeignet um Systeme zu vergleichen

Realität:

- Zellen sind keine Hexagons und auch nicht rund
- isotrope Ausbreitung untypisch
- exponentielle Dämpfung inadäquat
- Regelmäßige Standorte nicht machbar

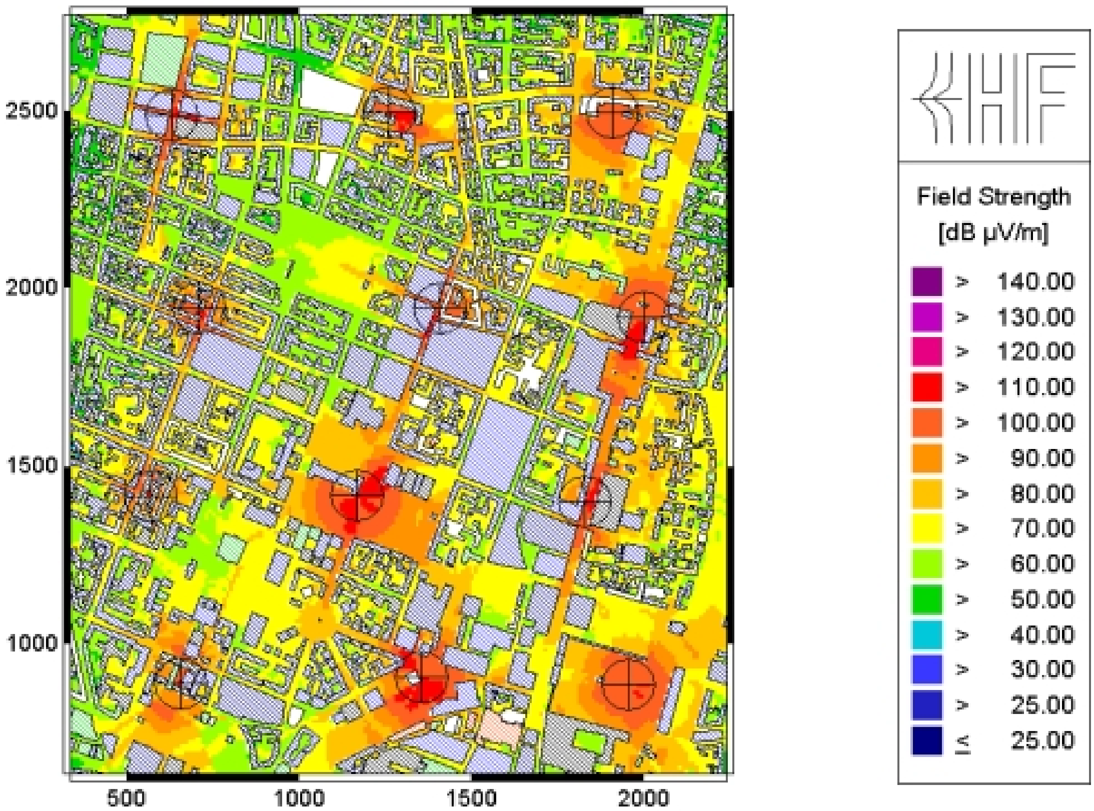
Weitere Probleme:

- Anpassung an unterschiedliche Verkehrsdichten
- zeitliche Schwankungen des Verkehrs
- unterschiedliche Zellgrößen

Realer Ansatz: Computergestützte Funknetzplanung

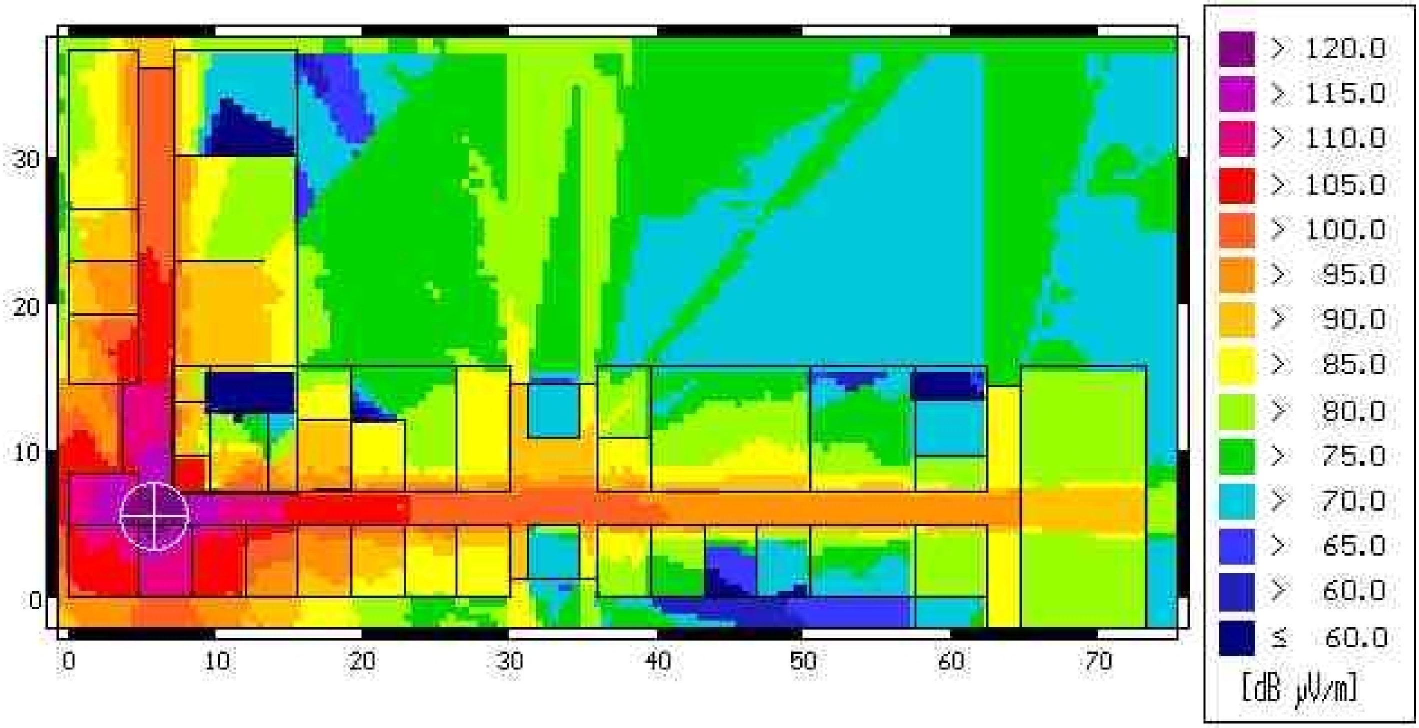
- **BS Planung**: Standort, Antennenhöhe, Antennencharakteristik, Sendeleistung
- **Zellulare Analyse**: Feldstärkeprädiktion, Zuordnungswahrscheinlichkeit, Interferenzanalyse
- Frequenzzuweisung

Radio Network Planning in Urban Environment

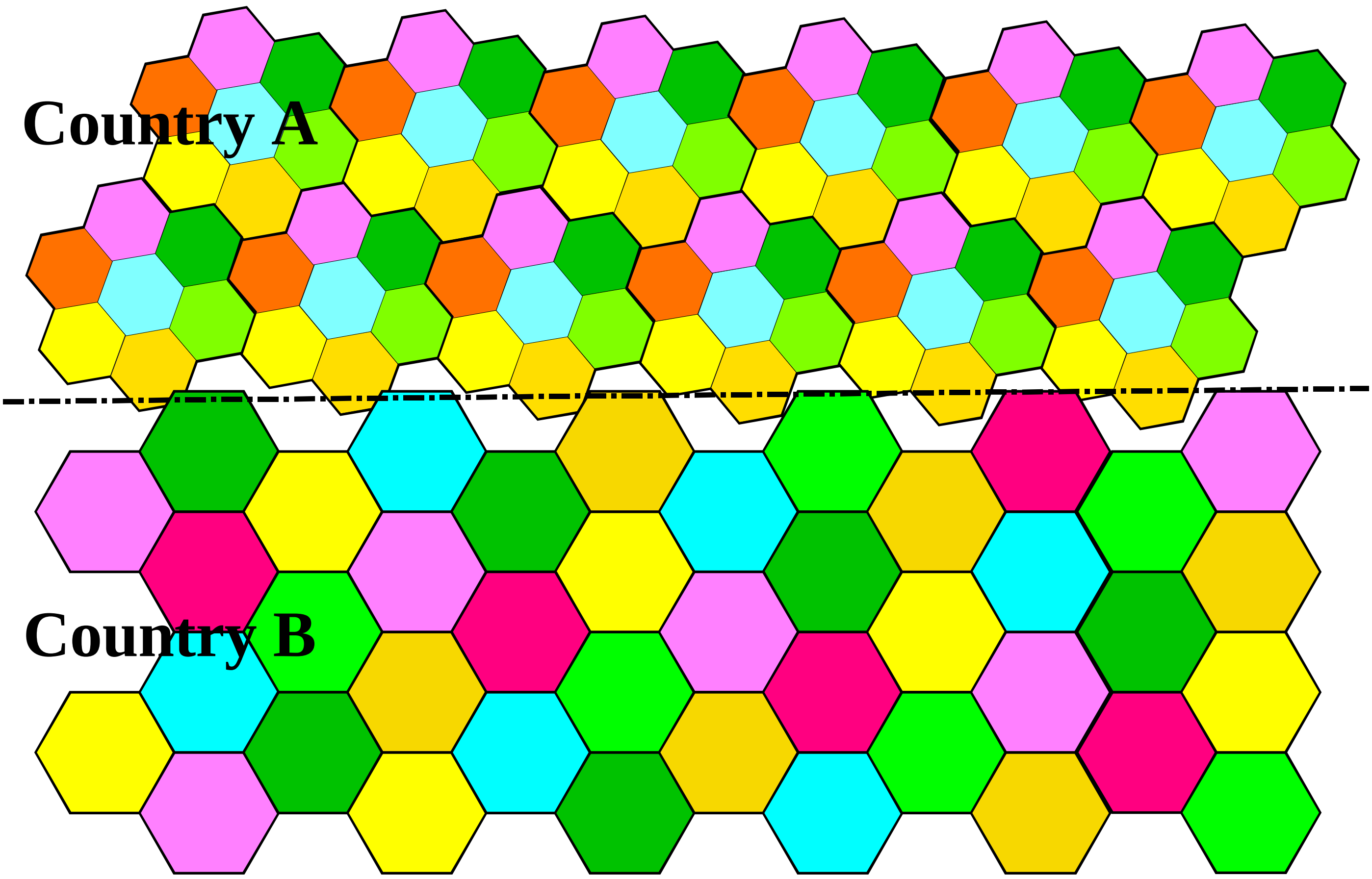


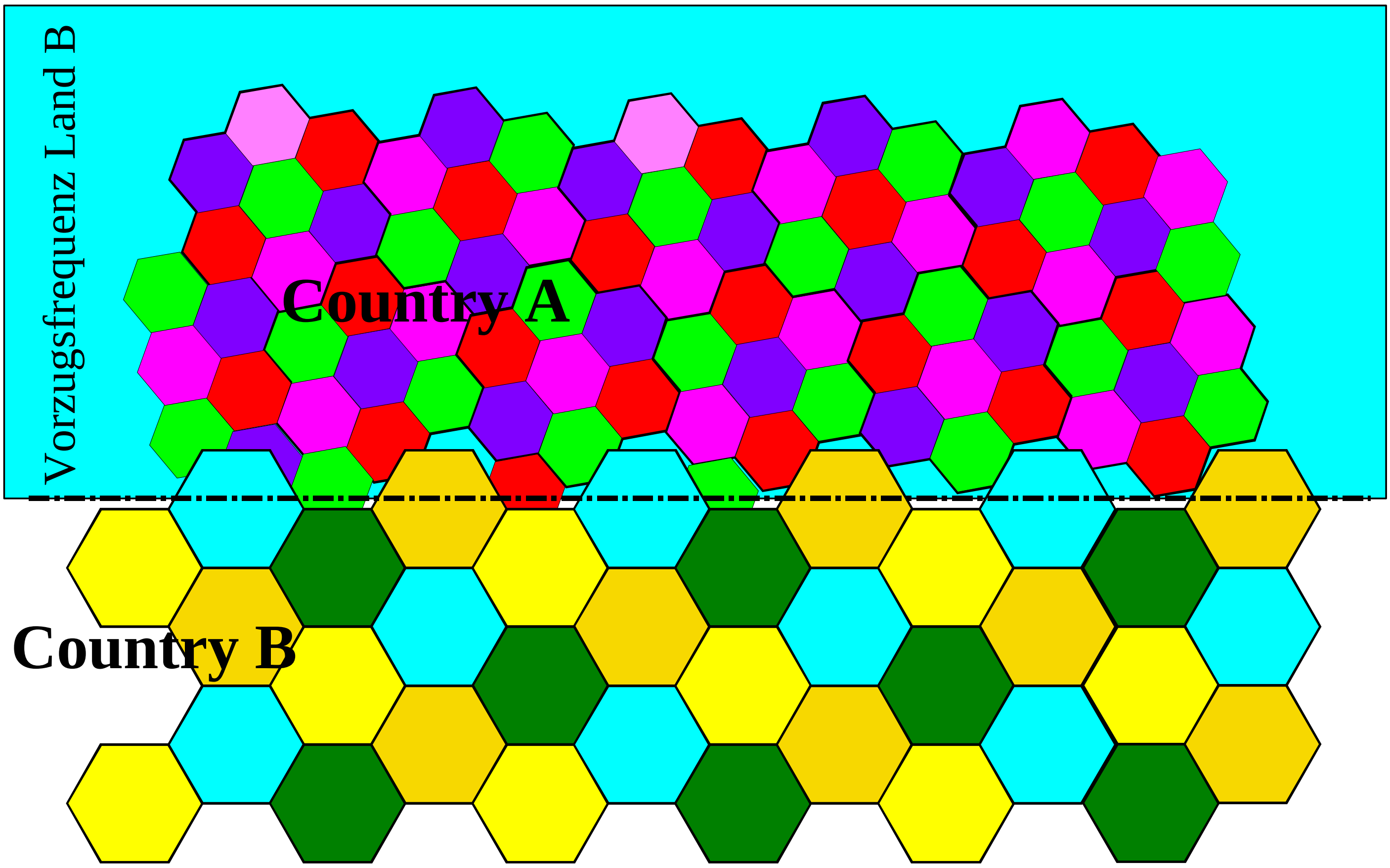
Optmiertes UMTS -Netzwerk in München Quelle: http://www.ihf.uni-stuttgart.de/ Institut für Hochfrequenztechnik

Radio Network Planning in Buildings



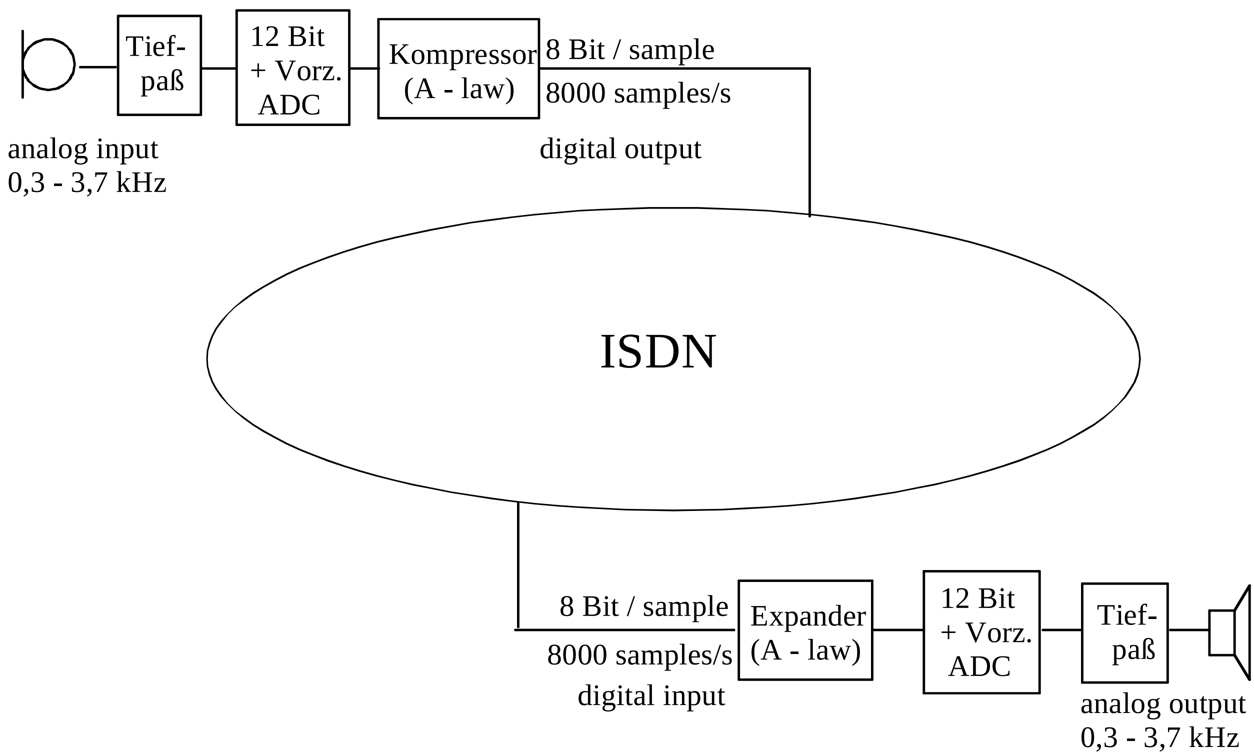
Prognose für eine Basisstation innerhalb eines Gebäudes im UHF-Frequenzband Quelle: http://www.ihf.uni-stuttgart.de/ Institut für Hochfrequenztechnik



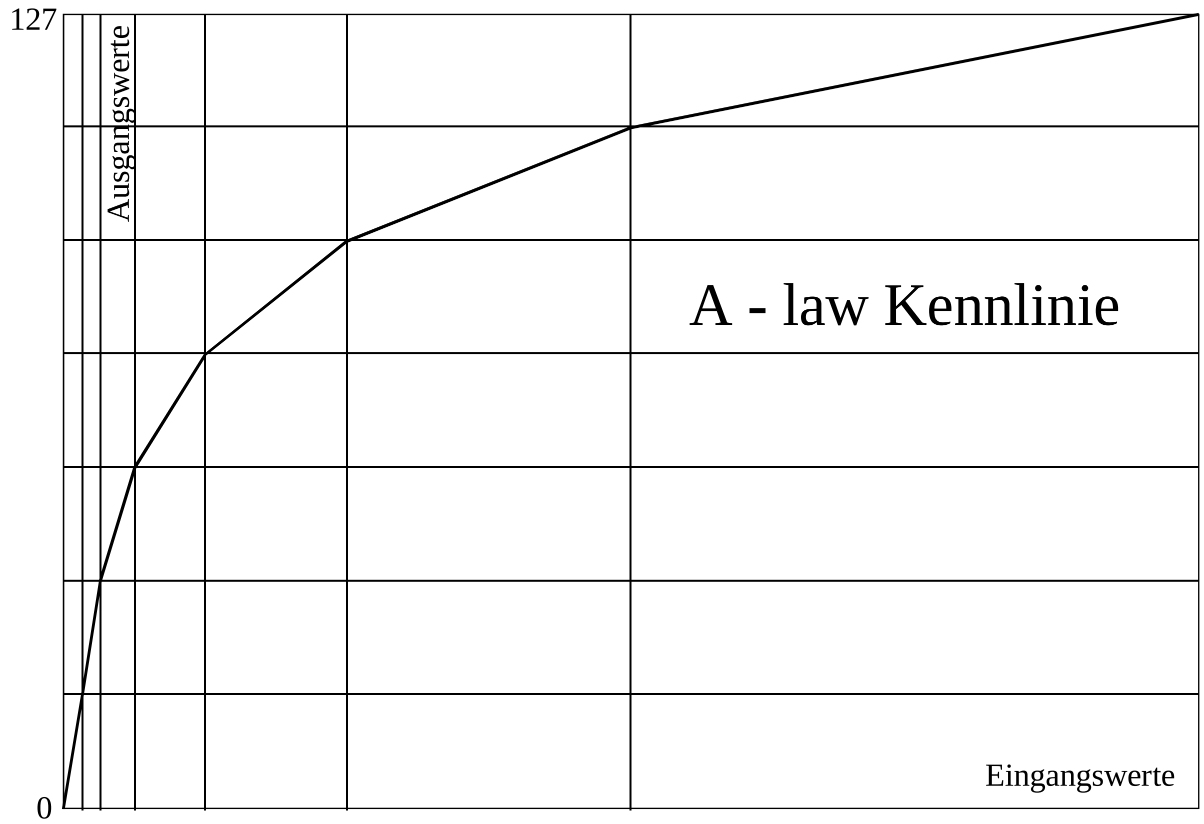


2 Speech Coding

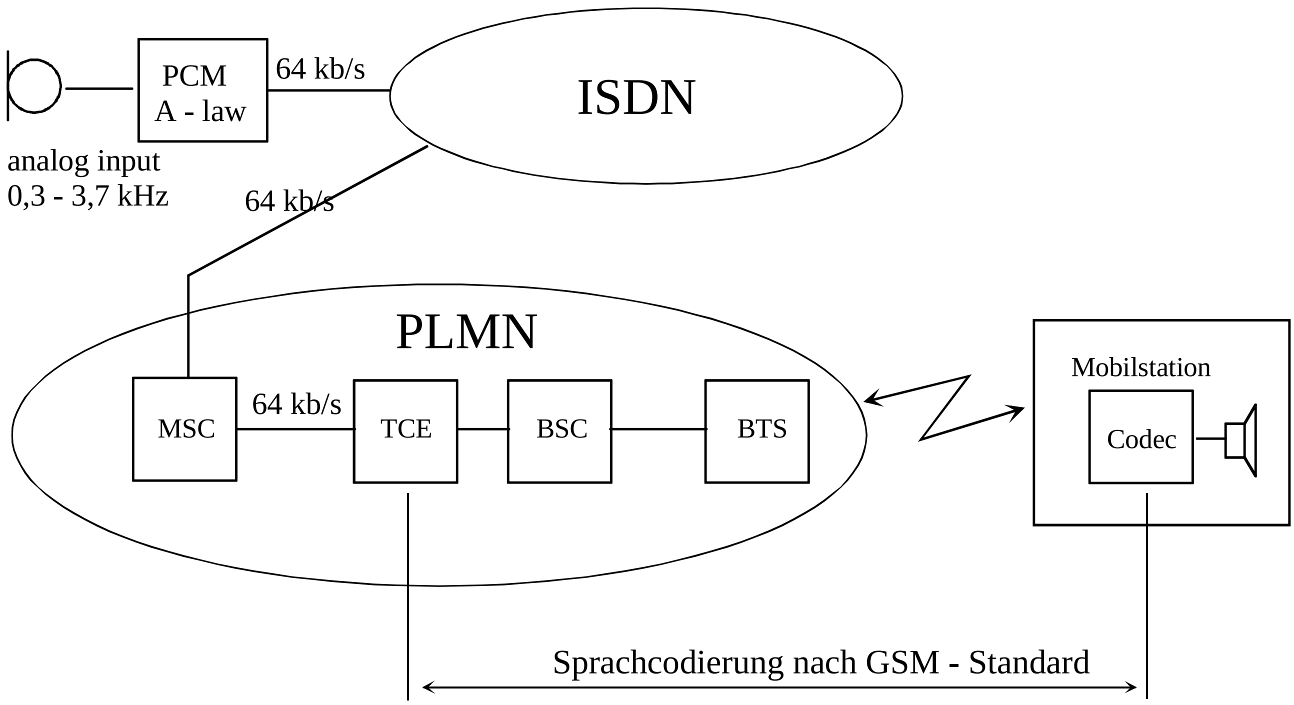
2.1 PCM A - law



Sprachcodierung : PCM A - law



Sprachcodierung : ISDN und Mobilfunk

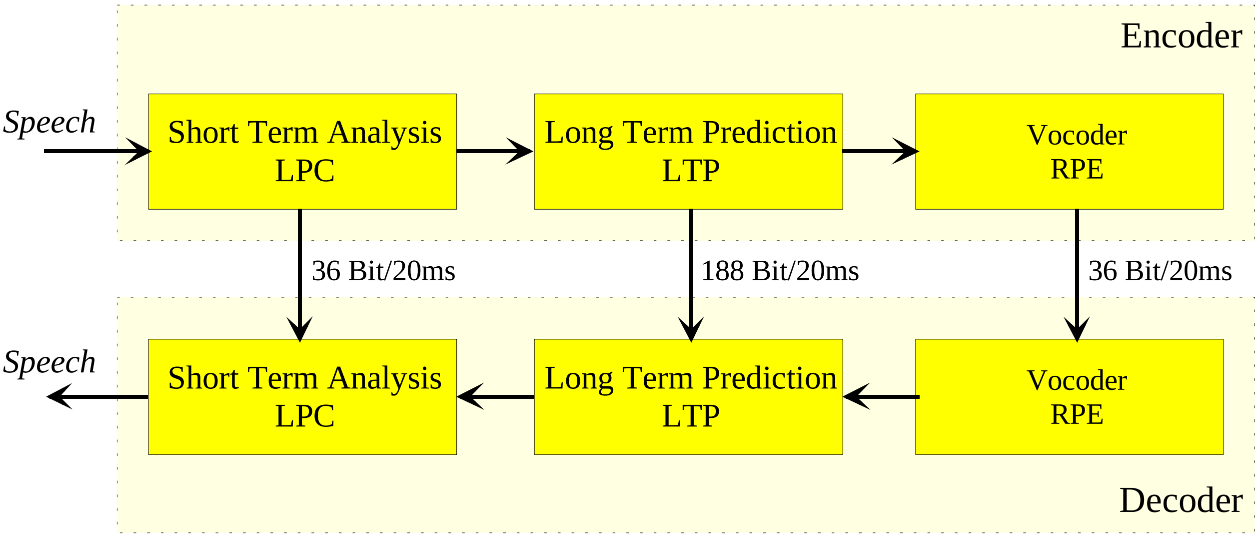


2.2 GSM Full Rate Codec: RPE - LTP

13 kb/s speech + 9,8 kb/s channel coding   net compression: 64/13 = 4,9

Hybrides Verfahren: Mischform aus *waveform encoding* und Vocoder Technik

- Ein Teil der Information wird als Abtastsignal übertragen: *waveform encoding* wie PCM, ADPCM
- Ein Teil der Information ist in einem Parametersatz codiert, der es dem Empfänger erlaubt, das Signal zu rekonstruieren (Vocoder Technik)



GSM Full Rate Codec: RPE - LTP

1. Sprachdatenvorverarbeitung:
   - 20 ms Daten sammeln = 160 Bytes = 1.280 Bit
   - Gleichanteil entfernen
   - Preemphasize-Filter hebt höhere Frequenzen im Spektrum an

2. LPC = Linear Predictive Coder (Kurzzeit Analyse):

   die Koeffizienten eines Filters werden berechnet, das die Einhüllende Amplitudenspektrums möglichst eben über der Frequenz macht. Die LPC Analyse berechnet die Koeffizienten des Filters

   Die Filterkoeffizienten (Reflexionskoeffizienten) werden logarithmiert übertragen (36 bit/20ms = 1,8 kb/s)

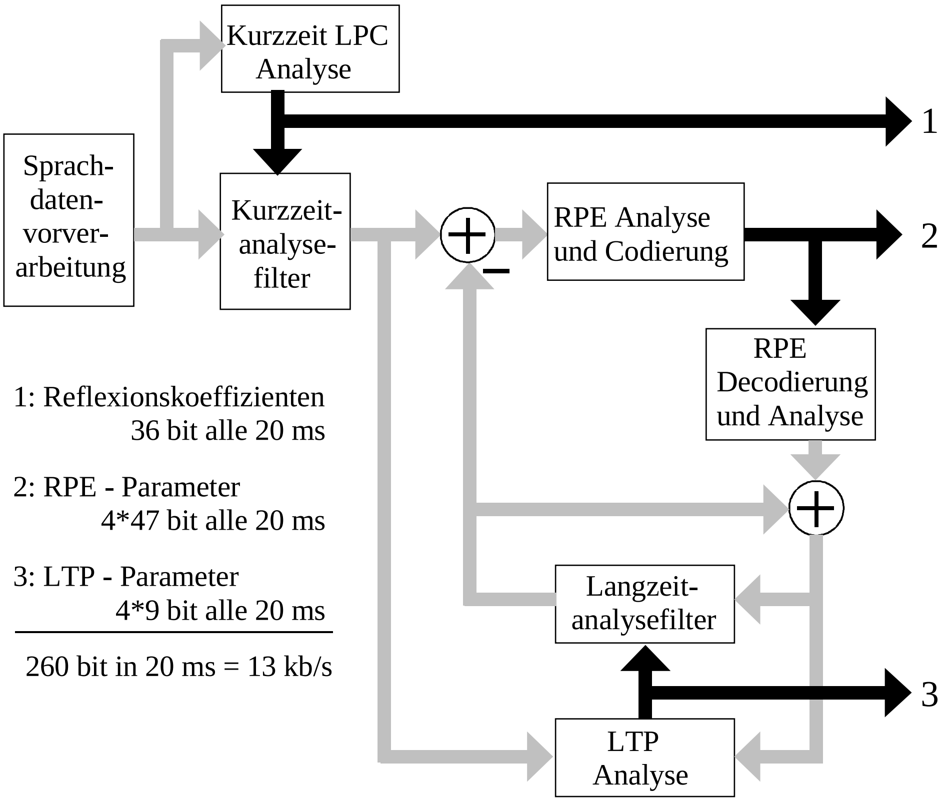
3. LTP = Long Term Prediction (Langzeit Analyse):

   die Koeffizienten des Langzeitanalysefilters werden berechnet und übertragen (4*9 bit/20ms = 1,8 kb/s)

4. RPE = Regular Pulse Excitation (Regular Pulse Excitation):

   Das Signal nach der LPC und LTP Filterung wird nach dem Vokoderverfahren analysiert.

GSM Full Rate Codec: RPE - LTP



1: Reflexionskoeffizienten
36 bit alle 20 ms

2: RPE - Parameter
4*47 bit alle 20 ms

3: LTP - Parameter
4*9 bit alle 20 ms

260 bit in 20 ms = 13 kb/s

LPC

Linear Predictive Coder

LTP

Long Term Prediction

RPE

Regular Pulse Excitation

2.3 Channel Coding for GSM Full Rate Codec

1. Die 260 Bits werden nach Relevanz in 3 Gruppen sortiert:

Gruppe 1 - höchste Relevanz = 50 Bits
Gruppe 2 - mittlere Relevanz = 132 Bits
Gruppe 3 - niedrige Relevanz = 78 Bits

2. Parity Bits zur Fehlererkennung für die Gruppe 1 einfügen

hohe Relevanz 50 Bit	Parity 3 Bit	mittlere Relevanz 132 Bit	niedrige Relevanz 78 Bit

3. Kanalcodierung der Gruppe 1 und 2
Es werden 4 Tailbits angehängt

hohe Relevanz 50 Bit	Parity 3 Bit	mittlere Relevanz 132 Bit	Tail 4 Bits

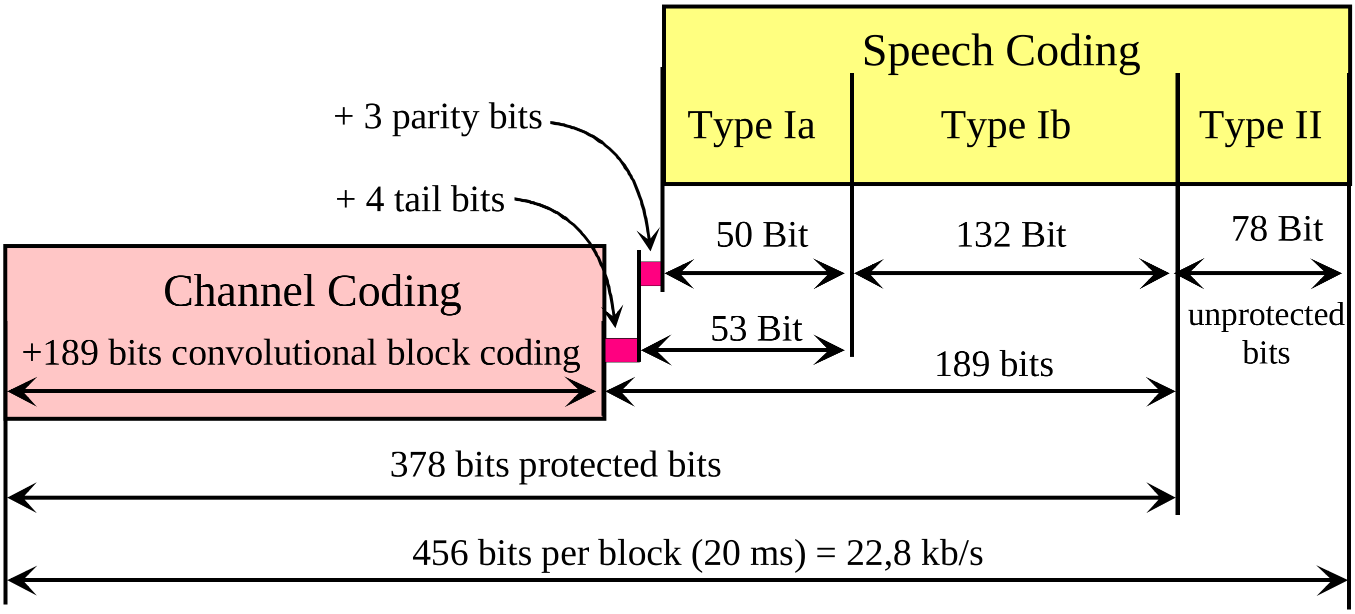
= 189 Bits

½ Rate Convolutional Coding
378 Bits coded + 78 Bit uncoded = 456 Bit

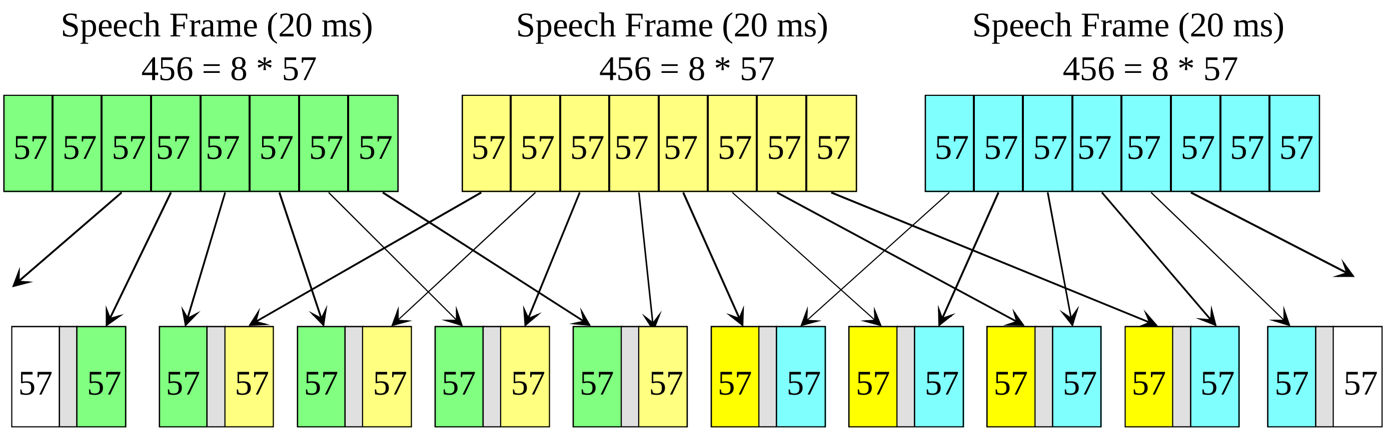
hohe und mittlere Relevanz 189 Bit Info + 189 Bit Redundanz = 378 Bits	niedrige Relevanz 78 Bit

4. Übertragungsrate: 456 Bit in 20 ms ➔ 22,8 kb/s

GSM Full Rate Codec: Channel Coding



Channel Coding for Speech: Interleaving



2.4 Speech Delay

Full rate codec:

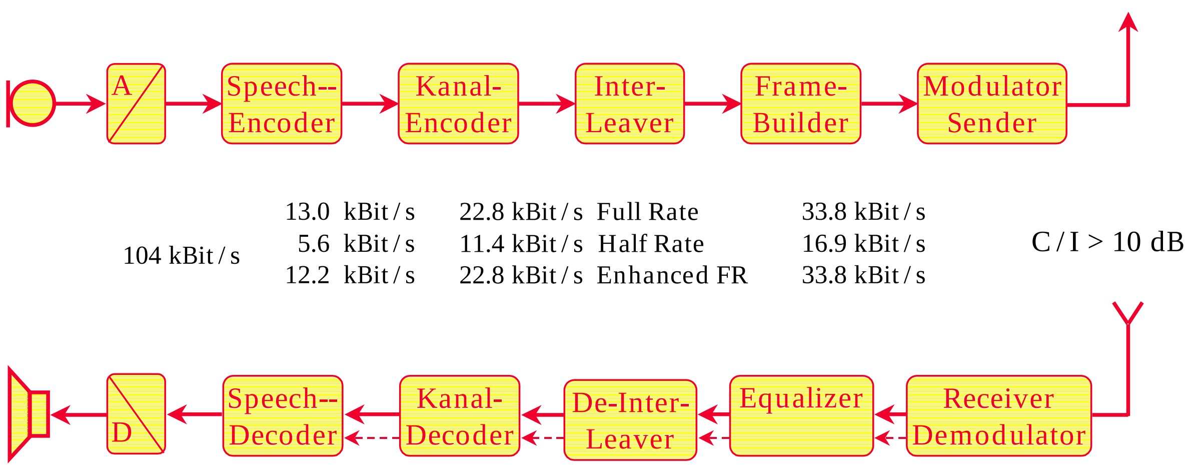
2 ms	AD-Wandler
20 ms	Segmentierung (Sprachblöcke von 20 ms)
3 ms	speech coder
17 ms	Übertragung über 16 kb/s (Abis)
2 ms	Vermittlung
1 ms	channel coding
37 ms	interleaving U_m
4 ms	Demodulator
0 ms	de-interleaving
3 ms	channel decoder
1 ms	speech decoder
2 ms	DA-Wandler
-------	
92 ms	

2.5 GSM Speech Transmission

first generation of speech codecs

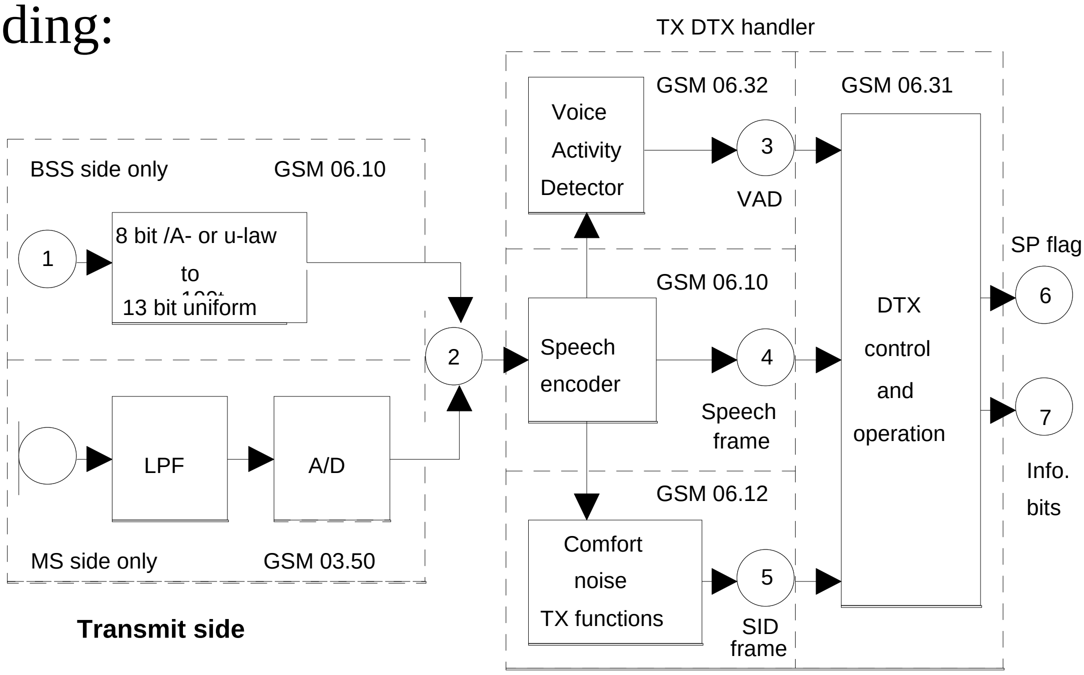
13 kb/s	FR	Full Rate Codec
5,6 kb/s	HR	Half Rate Codec
12.2 kb/s	EFR	Enhanced Full Rate Codec

Optimized for C/I = 10 dB



Speech Coding:

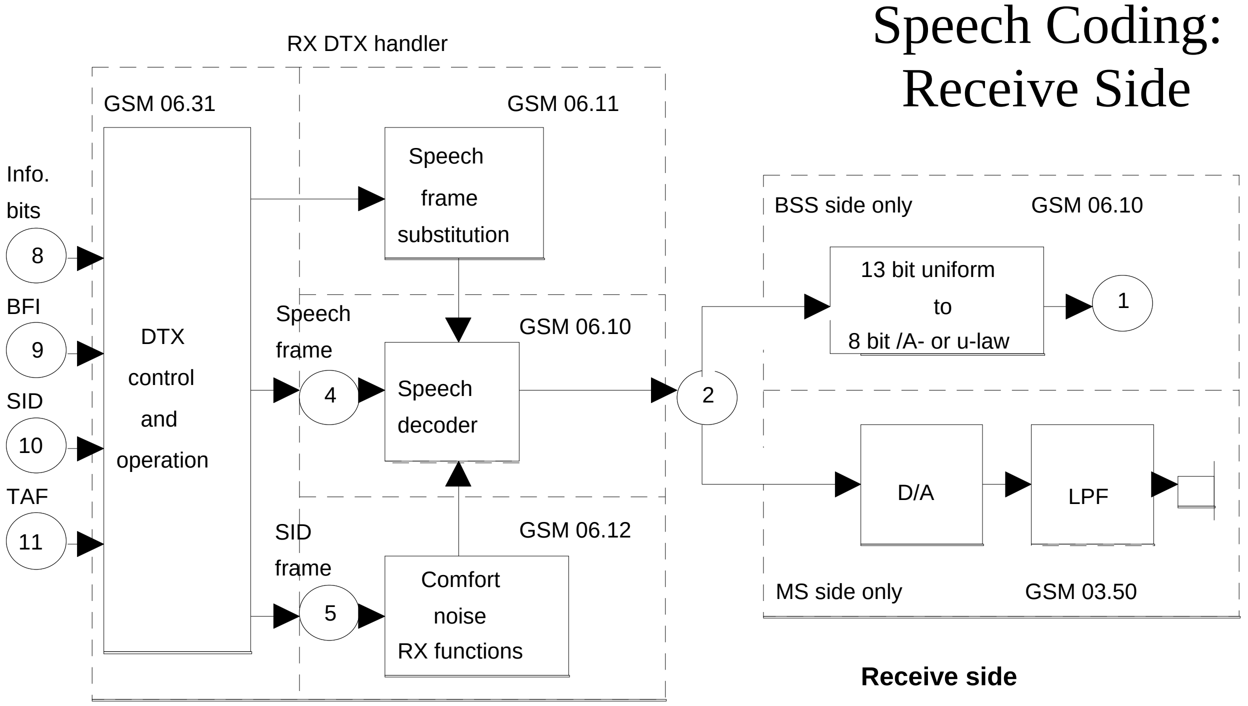
Transmit Side



Transmit side

(1) 8 bit /A- or u-law (PCS 1900) PCM (ITU-T rec G.711) 8000 samples/s
(2) 13 bit uniform PCM, 8000 samples/s
(3) Voice activity flag
(4) Encoded speech frame, 50 samples/s, 260 bits/frame
(5) Silence Descriptor (SID) frame, 260 bits/frame
(6) Speech flag, indicates whether information bits are speech or SID information
(7) Information bits delivered to the radio subsystem

Speech Coding: Receive Side



Receive side

(1) 8 bit /A- or u-law (PCS 1900) PCM (ITU-T rec G.711) 8000 samples/s
(2) 13 bit uniform PCM, 8000 samples/s
(4) Encoded speech frame, 50 samples/s, 260 bits/frame
(5) Silence Descriptor (SID) frame, 260 bits/frame

(8) Information bits received from the radio subsystem
(9) Bad Frame Indication (BFI) flag
(10) Silence Descriptor (SID) flag
(11) Time Alignment Flag (TAF), marks the position of the SID frame within the SACCH multiframe

2.6 Speech Quality

Mean Opinion Score MOS

Wert	Qualität	Verschlechterung der Qualität
5	sehr gut (excellent)	Nicht erkennbar
4	gut (good)	Erkennbar, aber nicht störend
3	akzeptabel (fair)	Erkennbar und leicht störend
2	schlecht (poor)	Störend, aber nicht unangenehm
1	sehr schlecht (bad)	Störend und sehr unangenehm

Beschreibung
5 - 4: Toll Quality
4 - 3: Communication Quality
3 - 1: Synthetic Quality

Speech Delay

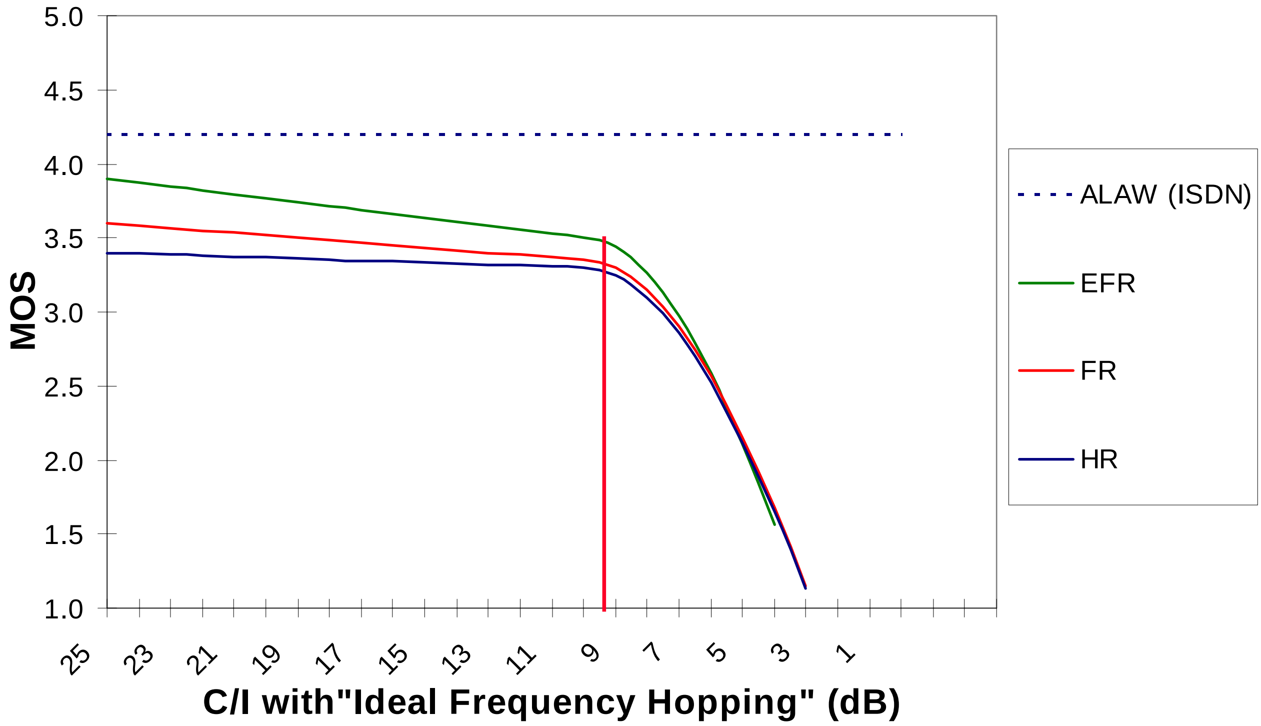
Verzögerung	Beschreibung (ITU Delay Recommendations)
0 – 150 ms	Akzeptabel für die meisten Anwendungen
150 – 400 ms	Akzeptabel unter der Berücksichtigung des Einflusses der Übertragungszeit auf die Übertragungsqualität der Anwendung
> 400 ms	Inakzeptabel für die allgemeine Netzwerkplanung; jedoch in Ausnahmefällen für spezielle Anwendungen toleriert

Codecs for Voice over IP

Codierungsverfahren	Bitrate (kbit/s)	Mean Opinion Score MOS	Coding Delay (ms)
G.711 PCM	64	4,10	0,75
G.726 ADPCM	32	3,85	1
G.728 LD-CELP	16	3,61	3-5
G.729 CS-ACELP	8	3,92	10
G.729a CS-ACELP	8	3,70	10
G.723.1 MP-MLQ	6,3	3,98	30
G.723.1 ACELP	5,3	3,65	30

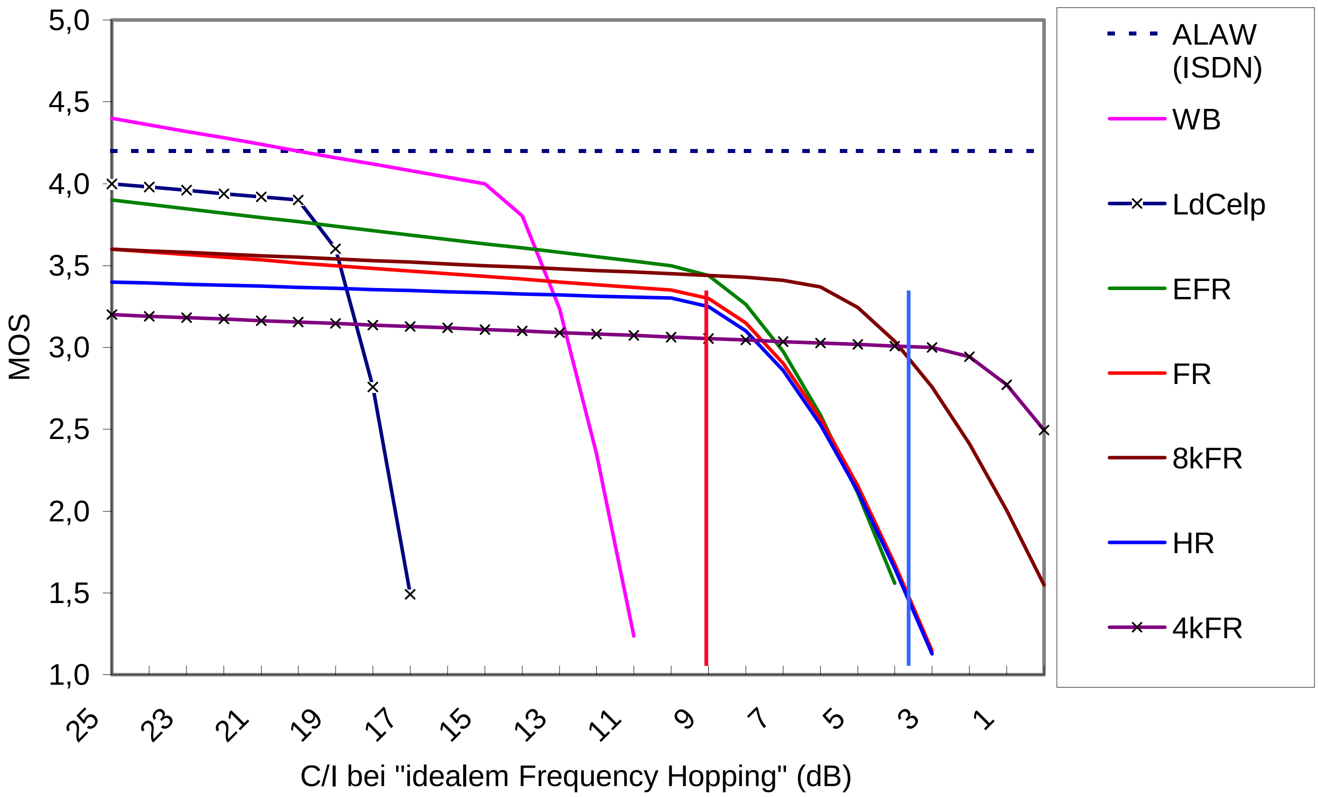
GSM Speech Quality

first generation of speech codecs: FR, HR, EFR



Speech Quality of various Codecs

AMR brings up to 6 dB higher robustness to GSM



3 AMR Adaptive Multi-Rate Codec

Optimization Goals

- Higher Capacity when possible ➔ Soft Capacity
- Better Speech Quality where possible
- Higher Robustness where needed

Codec-Family with high Flexibility
with fast Adaptivity
for GSM Full- and Half- Rate Channels

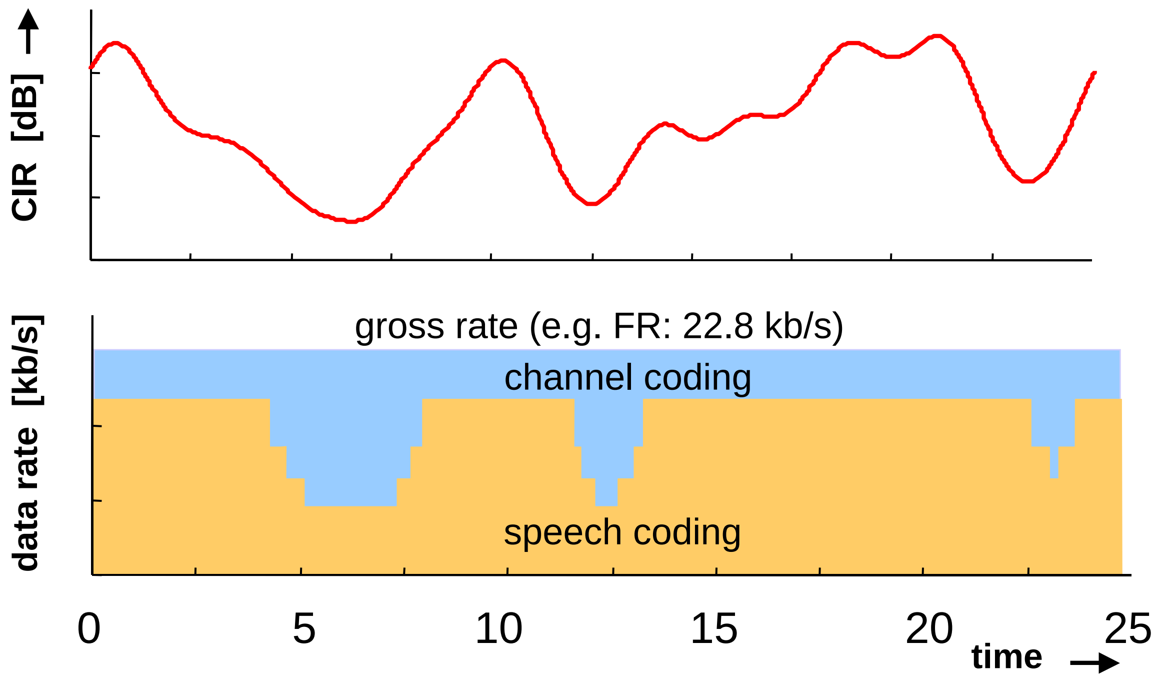
3.1 AMR Adaptive Multi-Rate Codec

GSM, UMTS, DAMPS, PDC

Coder Rate	UMTS	GSM Full Rate	GSM Half Rate	D-AMPS	
12.2 kBit/s	yes	yes			
10.2 kBit/s	yes	yes			
7.95 kBit/s	yes	yes	yes		
7.40 kBit/s	yes	yes	yes		EFR in D-AMPS
6.70 kBit/s	yes	yes	yes	yes	EFR in PDC
5.90 kBit/s	yes	yes	yes	yes	
5.15 kBit/s	yes	yes	yes	yes	
4.75 kBit/s	yes	yes	yes	yes	

Adaptive Multi-Rate Codec (AMR)

dynamic distribution between speech coding and channel coding



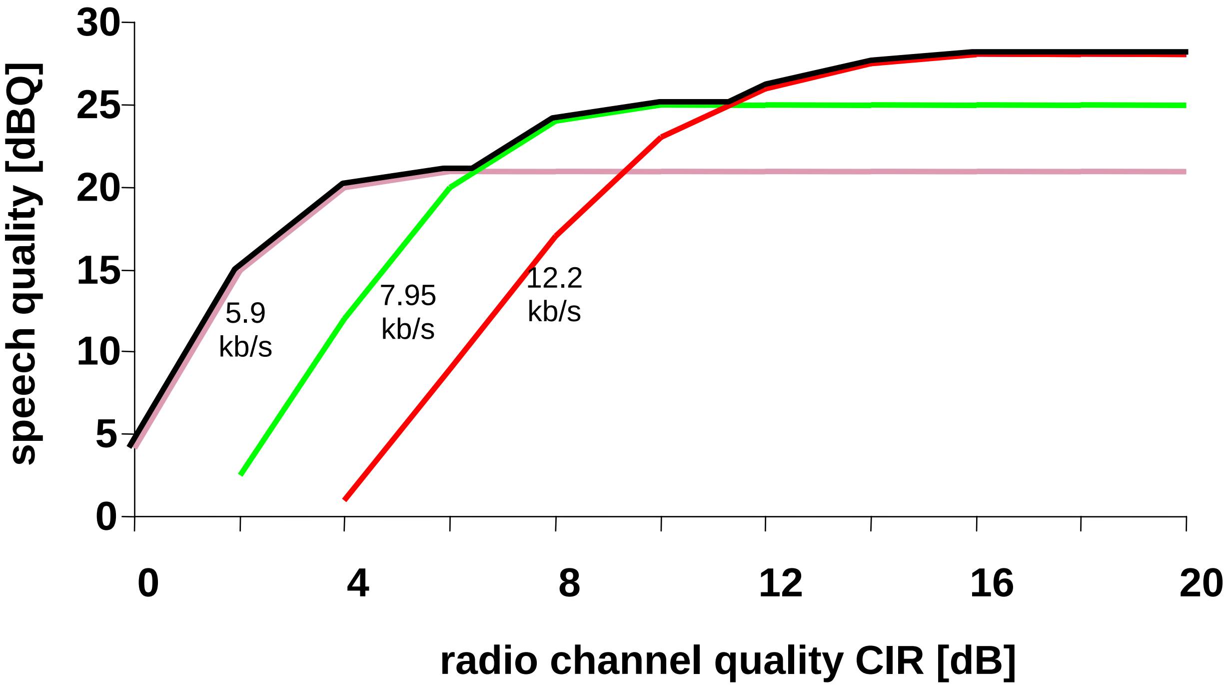
AMR in GSM

coding rate depends an channel quality

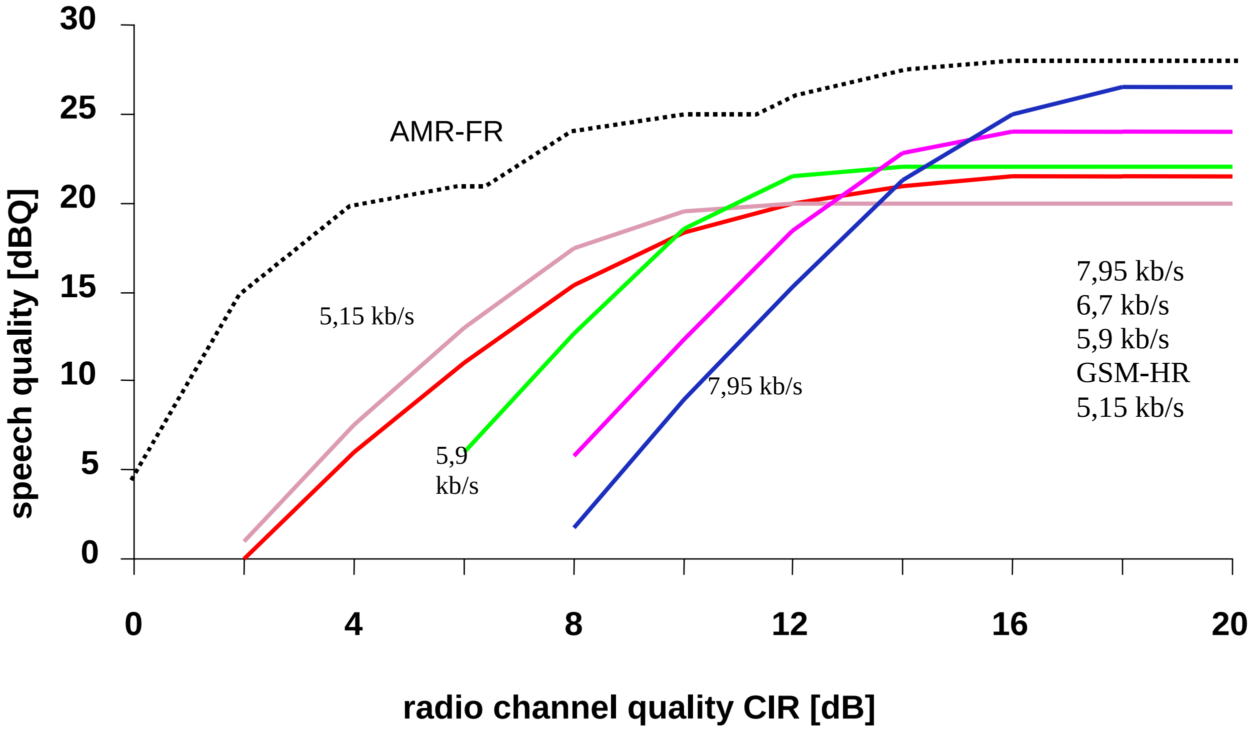
	GSM Full Rate		GSM Half Rate	
Speech Codeing Rate	Channel Codeing Rate	Transmission Rate	Channel Codeing Rate	Transmission Rate
12.2 kBit/s	10.6 kb/s	22.8 kb/s		
10.2 kBit/s	12.6 kb/s	22.8 kb/s		
7.95 kBit/s	14.85 kb/s	22.8 kb/s	3.45 kb/s	11.4 kb/s
7.40 kBit/s	15.4 kb/s	22.8 kb/s	4.0 kb/s	11.4 kb/s
6.70 kBit/s	16.1 kb/s	22.8 kb/s	4.7 kb/s	11.4 kb/s
5.90 kBit/s	16.9 kb/s	22.8 kb/s	5.5 kb/s	11.4 kb/s
5.15 kBit/s	17.65 kb/s	22.8 kb/s	6.25 kb/s	11.4 kb/s
4.75 kBit/s	18.05 kb/s	22.8 kb/s	6.65 kb/s	11.4 kb/s

full rate / half rate depends on the present network traffic load

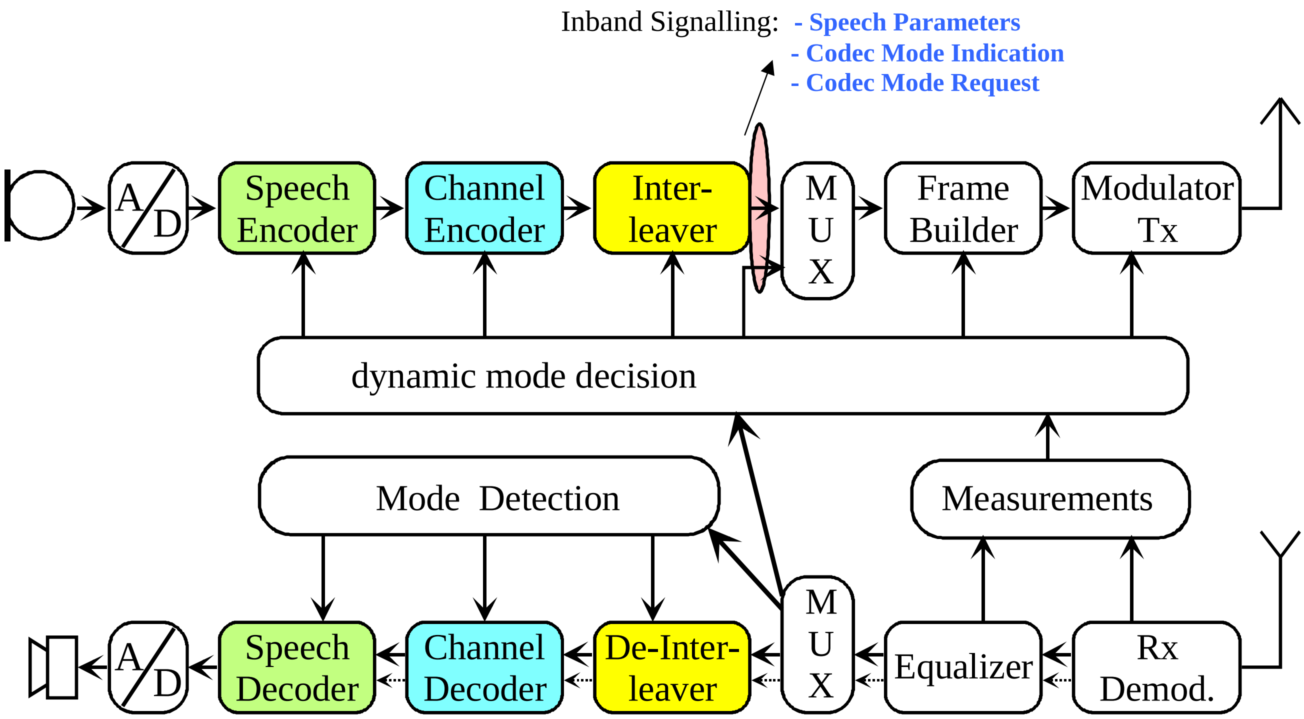
3.2 AMR Speech Quality on a Full Rate Channel



AMR Speech Quality on a Half Rate Channel



Adaptive Multi Rate Codec: Block Diagram



3.3 AMR Inband Signalling

Codec mode adaptation

- is fast, in order to be able to react on sudden changes of the radio channel conditions;
- is performed using inband signalling in the speech traffic channel as no sufficiently fast control channel exists in the GSM

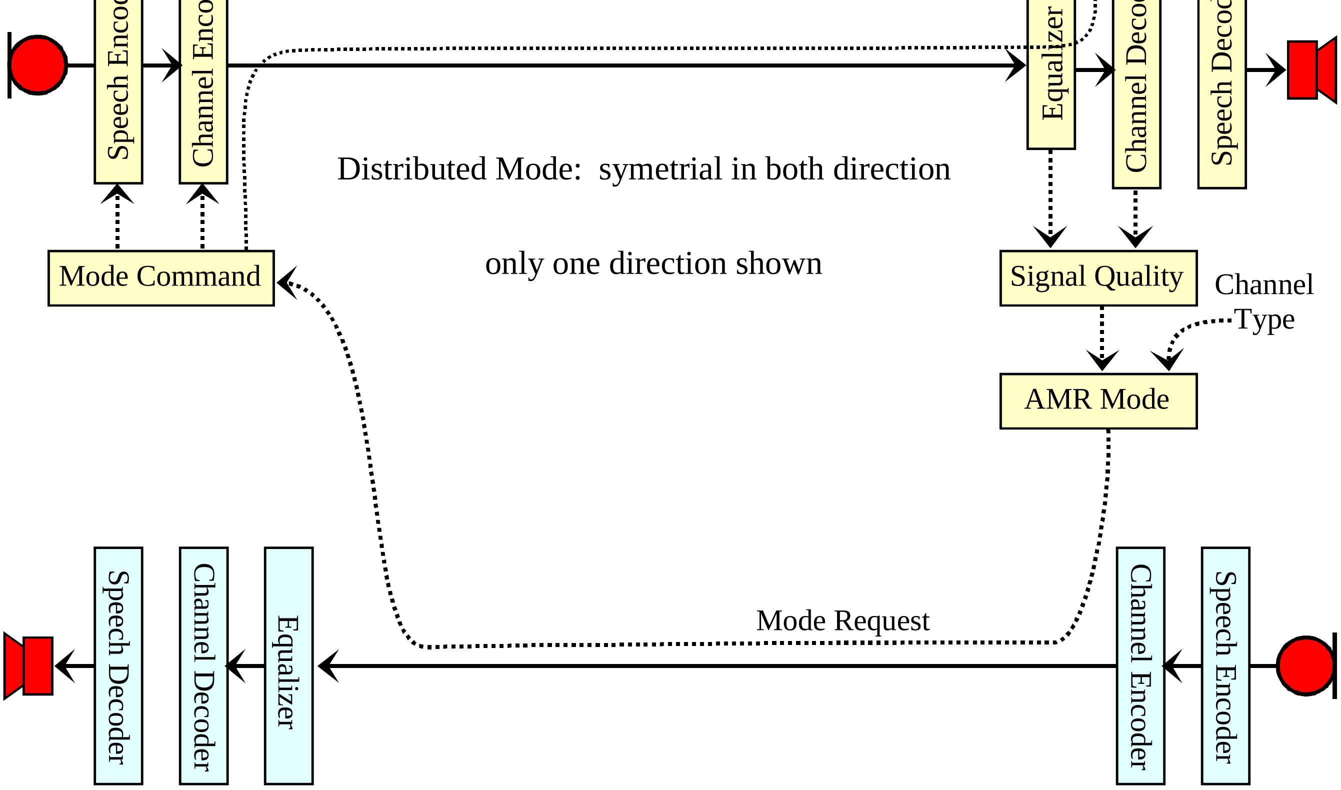
Fast, Parallel Signalling:

every speech frame:	- 2 net bits for	Codec Mode Indication	(the mode in use)
	- 2 net bits for	Codec Mode Request	(the mode to be used next)

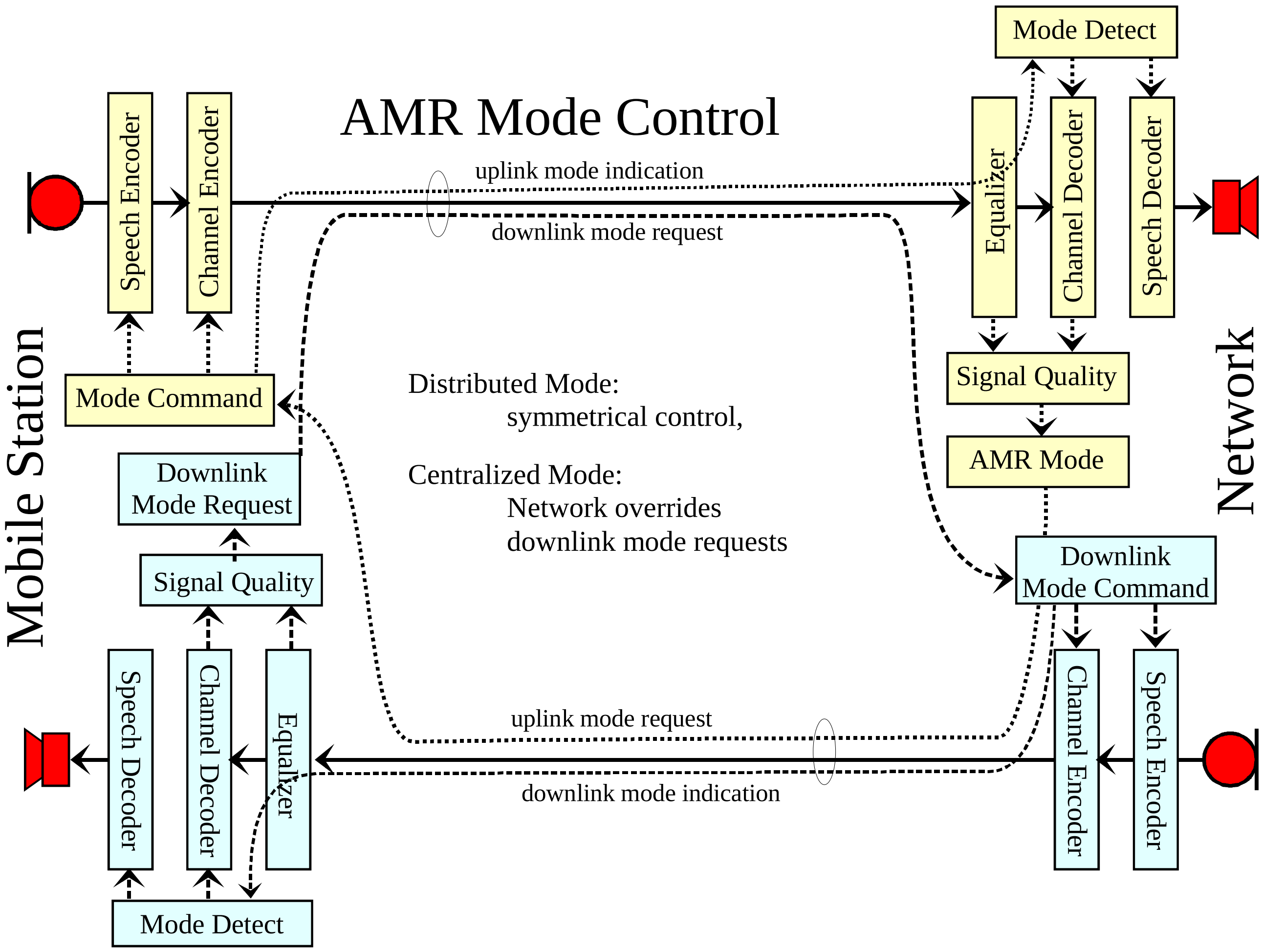
Slow, Robust Signalling: RATSCCH: frame stealing
(Robust AMR Traffic Synchronised Control Channel):

- Switch to Wideband-Modes ?
- Switch to “Speech Control” ?
  + Distributed Speech Recognition in uplink
  + Display and other feedback in downlink
- Switch to Data Modes?

AMR Mode Control



AMR Mode Control



AMR Mode Control

maximum speech quality at a given channel condition

Link Quality Measures require prediction to compensate for the adaptation loop delay

mode dependent measures are:

- the raw bit error rate (BER): re-encode the decoded sequence and compare it with the received gross bit sequence
- the residual BER,
- the frame erasure rate (FER), from the CRC of the most important speech bits

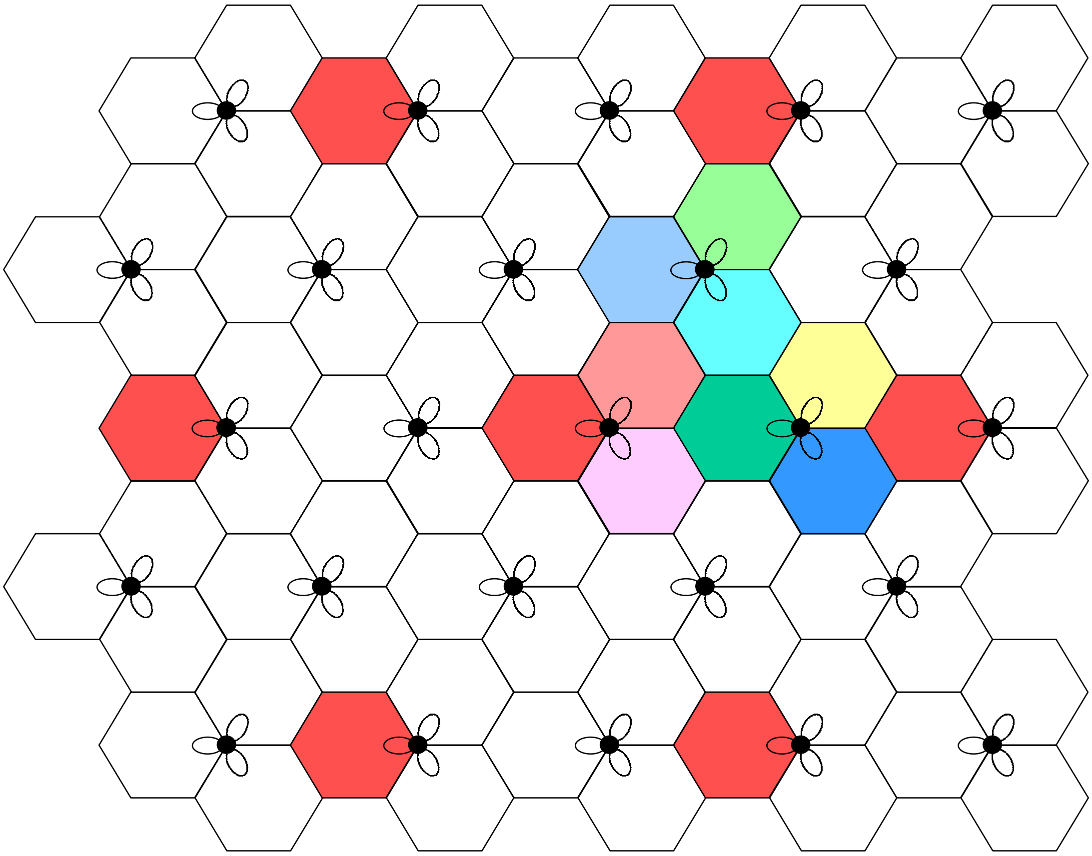
mode independent measures are:

- Carrier to Interference ratio (C/I) estimates derived from the Viterbi metric of the equalizer

3.4 Network Capacity with AMR Codec

- Assumptions (Model) :
  - Cell geometry — Hexagons
  - Frequency reuse — 9 (Cluster)
  - Path loss — $r^{-3.5}$
  - Shadowing — log-normal distribution; σ = 10 dB
  - Power control — due to received power (RxLev)
  - Allocation of a mobile to a cell — path loss
  - Subscriber distribution in the cell — equally distributed
  - Number of channels per cell — n = 32 Full Rate channels (4 TRX / cell)
  - Split between FR/HR — $k * FR + 2(n-k) HR; \quad 0 \leq k \leq n$
  - The number of FR and HR — k is fixed a simulation run
  - Offered traffic is adjusted to a blocking probability of 2%

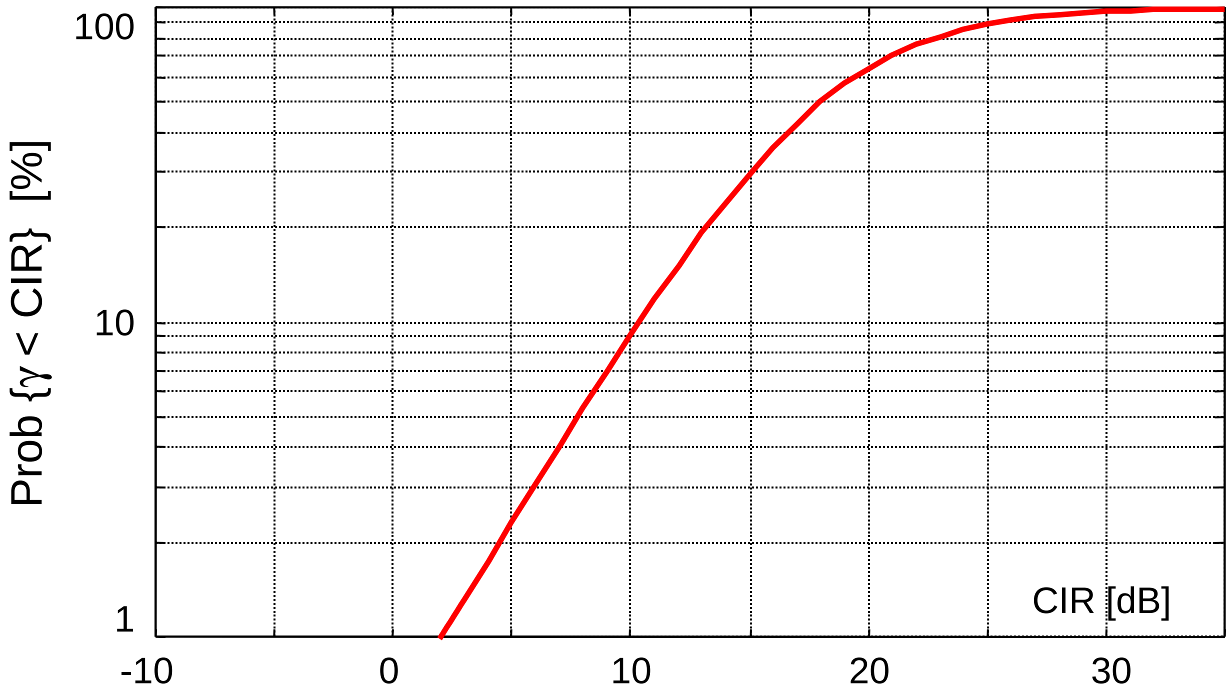
Sectorized Cells with Reuse 9



Estimation of Network Capacity with AMR Codec

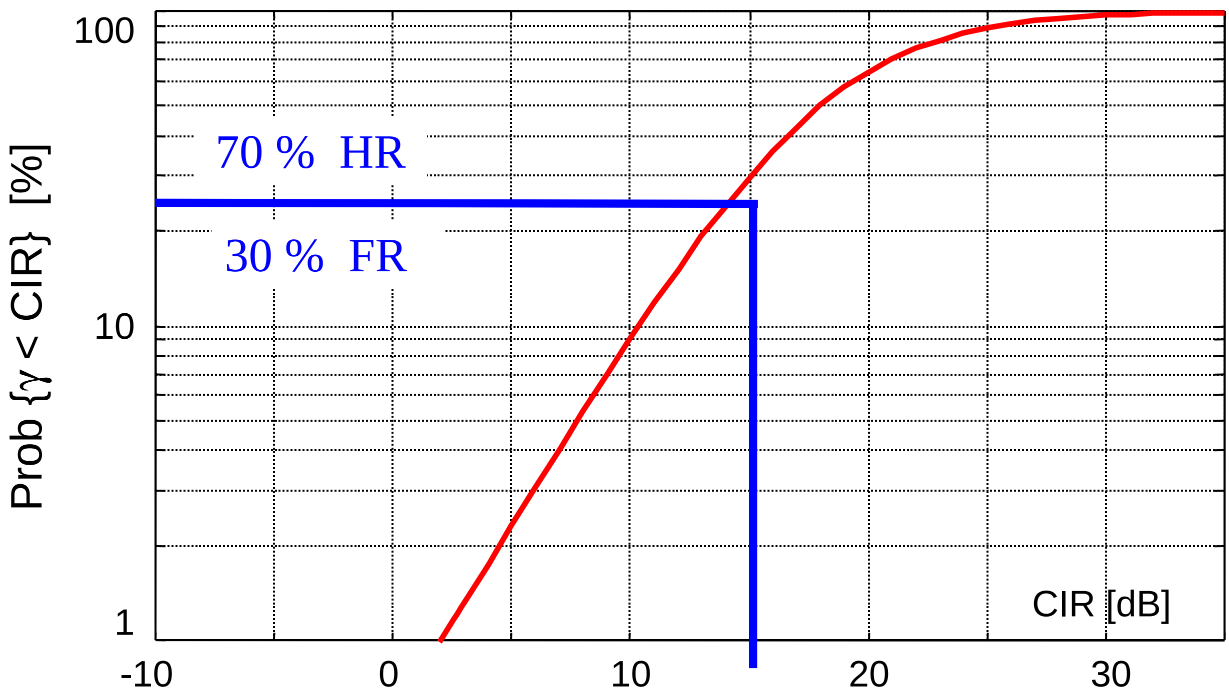
- Results :
  - For each connection:
    Path Loss ➔ CIR ➔ Codec selection ➔ Speech Quality
  - Probability Distribution of CIR
  - Probability Distribution of Quality

Result: CIR Distribution in a Cell



Shows the probability, that CIR of a connection γ is worst than a given CIR value
The figure depends only on radio network planning

Example: 30% FR channels and 70% HR channels

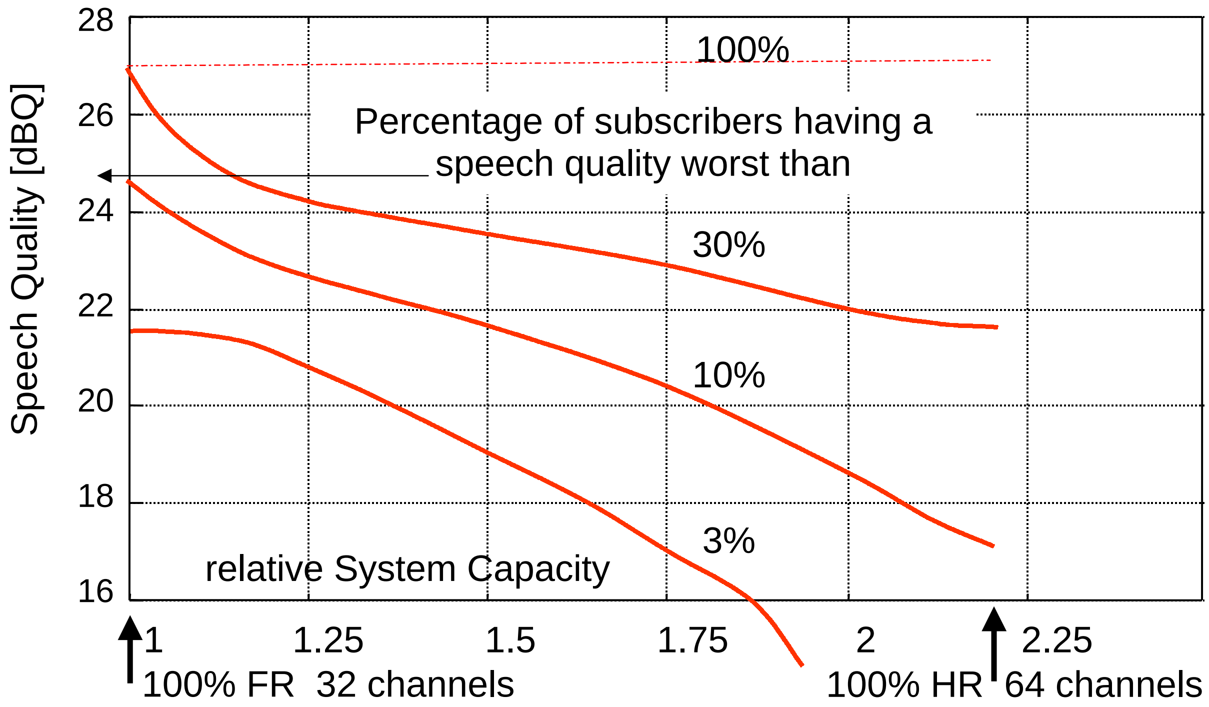


The subscribers with the worst CIR get a FR channel (30%), all others get a HR channel (70%)

Speech Quality vs. System Capacity

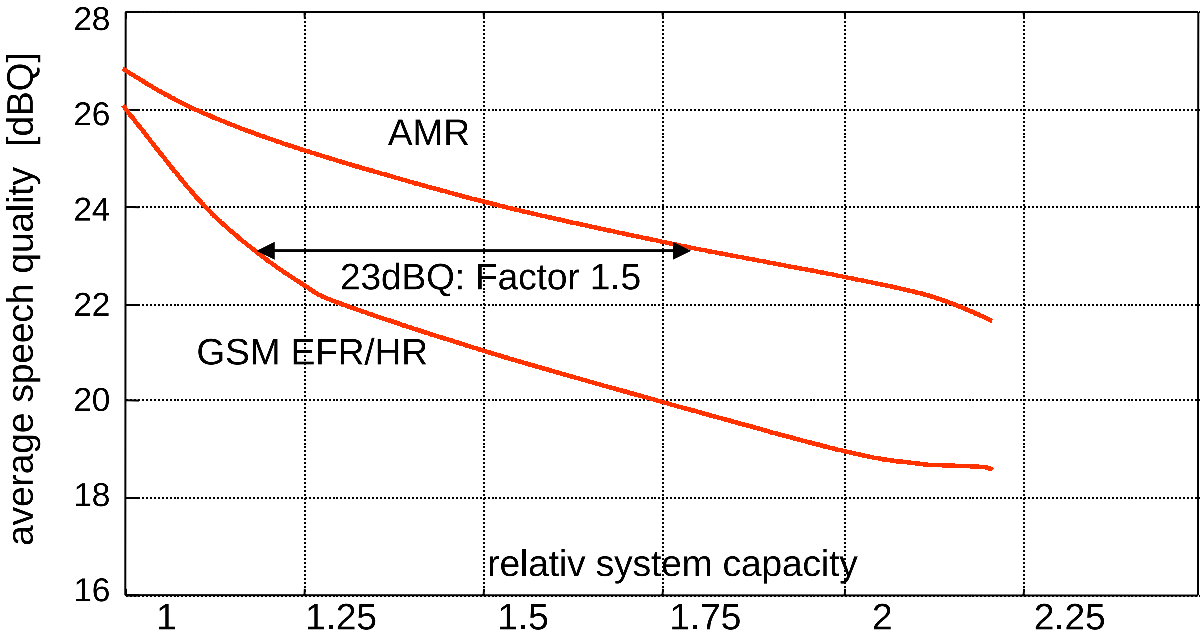
Half Rate ← → Full Rate

System capacity employing HR-codec is 2.2 compared to FR codec due to trunking gain



Speech Quality vs. System Capacity

EFR/HR ← → AMR

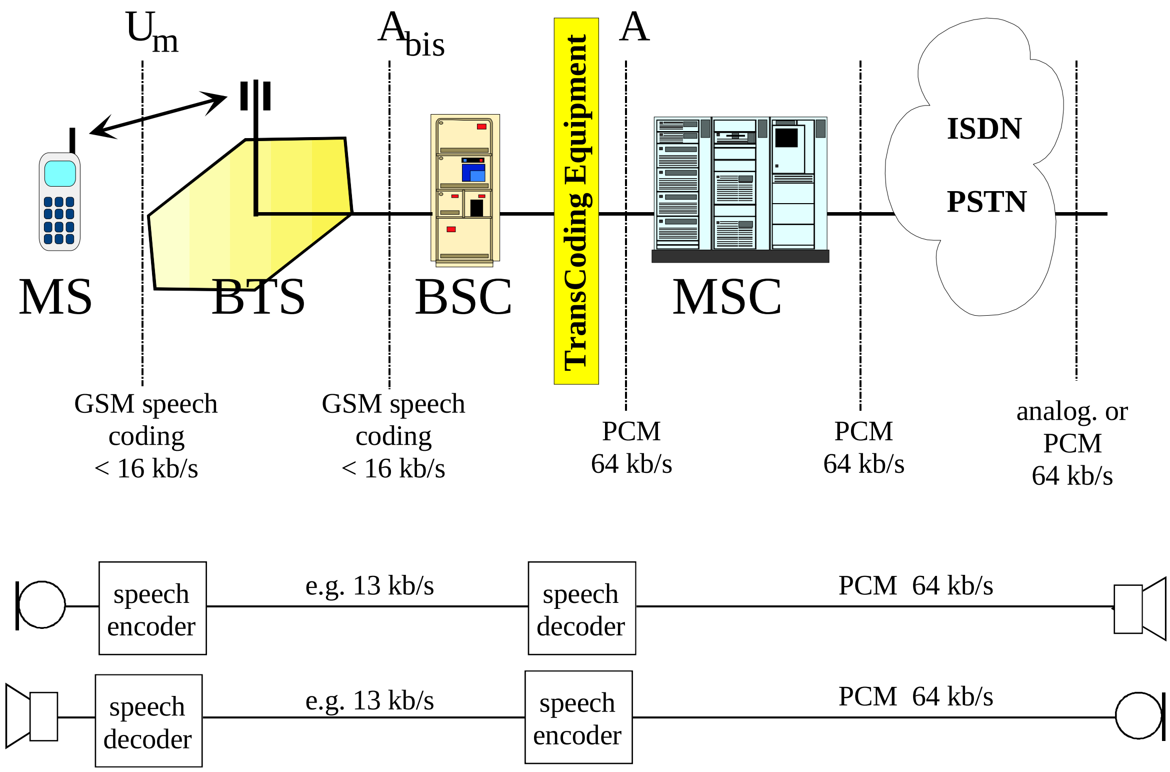


Example: If a speech quality of 23 dBQ is required, system capacity can be improved by a factor of **1.5** if AMR codec is employed instead of EFR/HR

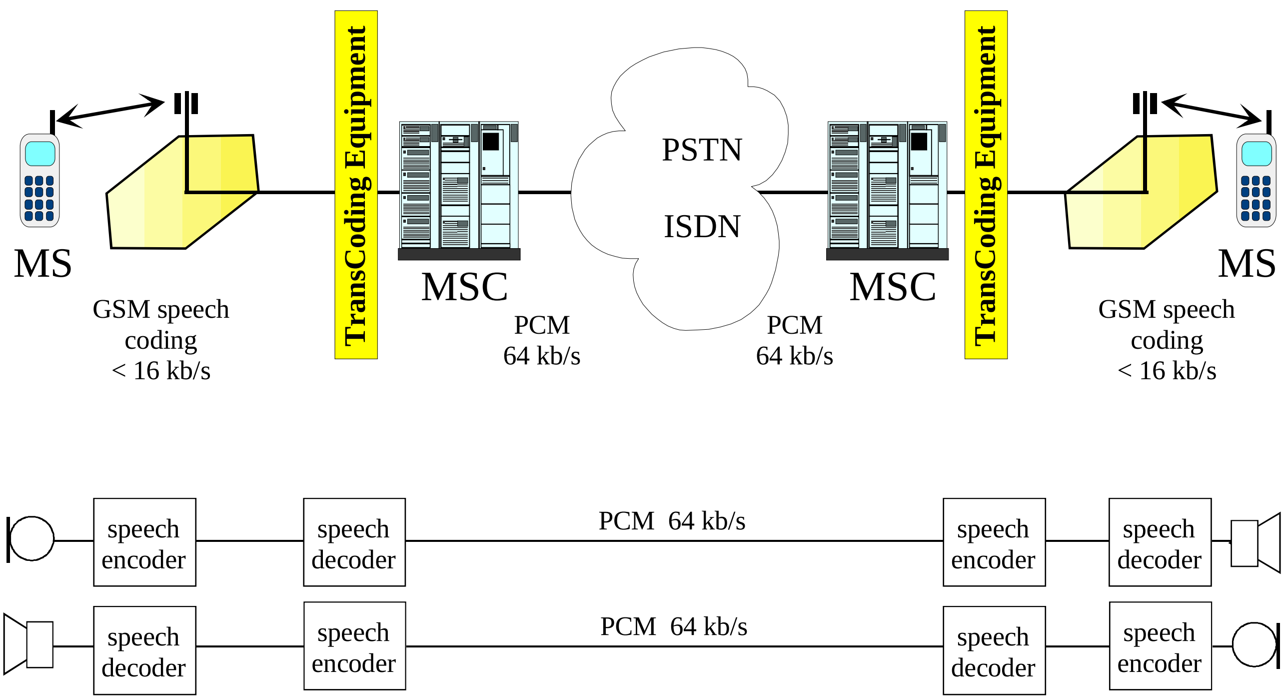
The curves show the average speech quality (mean value).

4 Tandem Free Operation

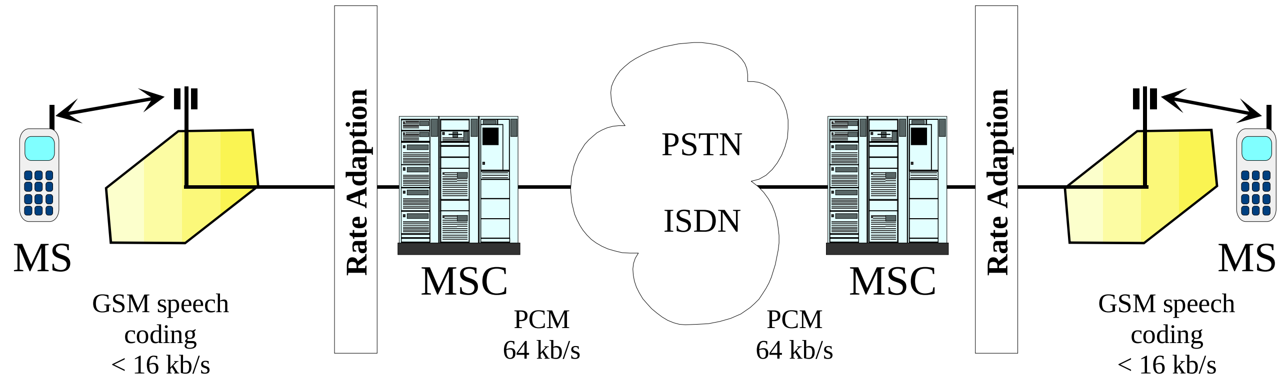
Speech Transmission: Fixed to Mobile

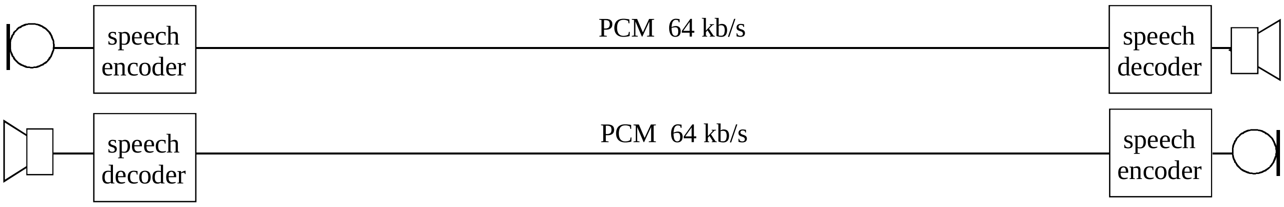


Mobile to Mobile Speech : Tandem



Mobile to Mobile Speech : Tandem Free





Why Tandem Free Operation?

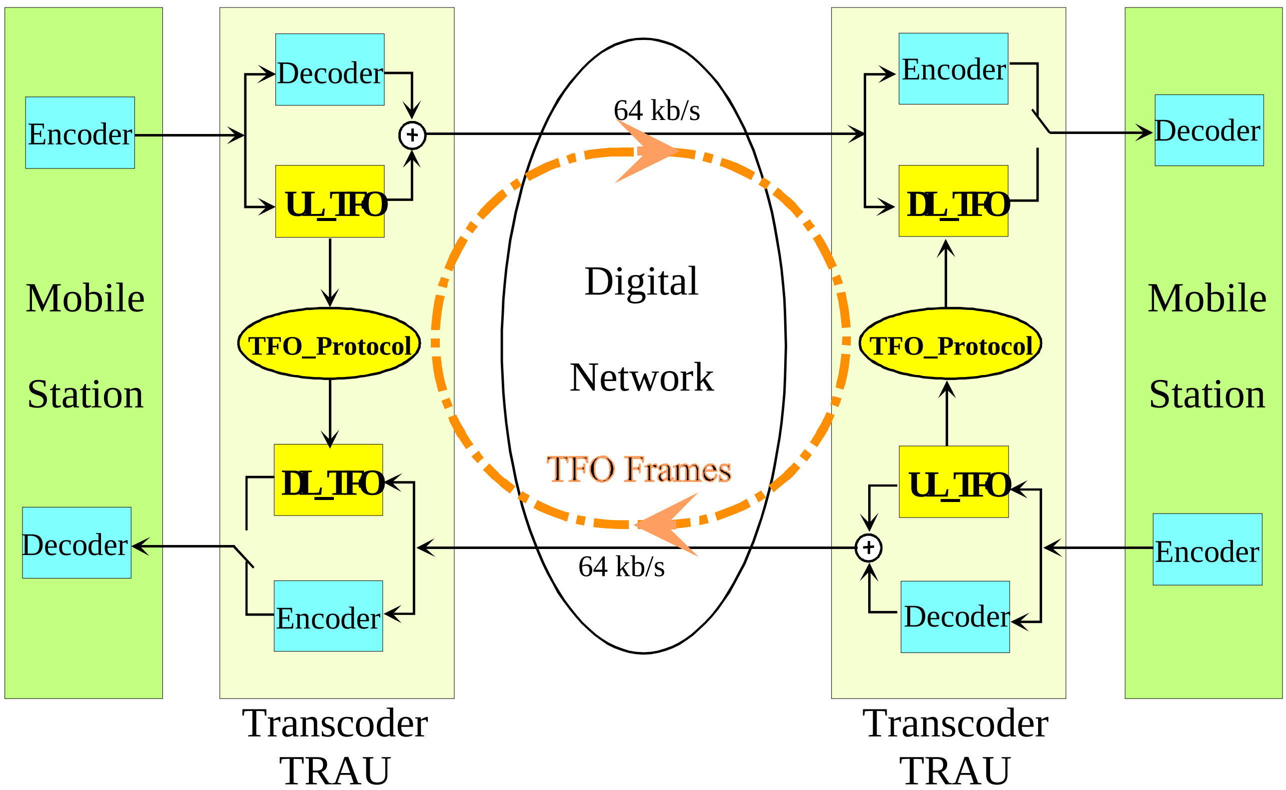
- Penetration of Mobiles increase rapidly
  => more and more mobile-mobile calls (majority in future)
- Speech Quality suffers from tandem transcoding:
  => 1) TFO is quality-optimal
- Transmission in fixed network not at 64 kBit/s, but at 8 or 16 kBit/s:
  => 2) TFO is cost-optimal
- Transcoder blocks digital data transmission:
  => 3) TFO is enabler for new applications

Vision: End-to-End-Compression, transparent network

Standardized Solution

- Inband signalling for TFO-Protocol in 64 kBit/s channel
  No Out-of-band signalling
- Only Transcoding Equipment (TRAU) affected;
  No other network entity involved.
- Generic Protocol for In-Path-Equipment (IPE);
  Codec in “hot standby” for fast fall back;
- Applicable / portable to all 2G and 3G Systems

TFO Architecture (Traffic Channel)



Inband Signalling in PSTN / ISDN traffic channel (64 kBit/s)

Replace in PCM coded speech the LSB of every 16th sample:

8 Bits of every sample	=	64.000 Bit/s
1 Bit of every sample	=	8.000 Bit/s
1 Bit of every 16th sample	=	**500 Bit/s**

=> 500 Bit/s embedded in 64 kb/s is practically inaudible

Header Sequence:	20 Bits
Command:	10 Bits
System Identification:	20 Bits
Every other extension block:	20 Bits

TFO Frame Format (16 kBit/s embedded in 64 kBit/s)

	Sample 1 (of 160)								Sample 9 (of 160)							
Bit 7	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Bit 6	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Bit 5	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Bit 4	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Bit 3	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Bit 2	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Bit 1	**S1**	**S2**	**S3**	**S4**	**S5**	**S6**	**S7**	**S8**	**C1**	**C3**	**C5**	**C7**	**C9**	**C11**	**C13**	**C15**
Bit 0	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**C2**	**C4**	**C6**	**C8**	**C10**	**C12**	**C14**

The upper six bits contain the decoded speech for “Fall Back” = PCM format.

The lower two bits contain the speech parameters embedded into the PCM samples.

Speech Frame Format (13 kBit/s embedded in 16 kBit/s)

Octet no.	Bit number 1	2	3	4	5	6	7	8
0	0	0	0	0	0	0	0	0
1	0	0	0	0	0	0	0	0
2	1	C1	C2	C3	C4	C5	C6	C7
3	C8	C9	C10	C11	C12	C13	C14	C15
4	1	D1	D2	D3	D4	D5	D6	D7
5	D8	D9	D10	D11	D12	D13	D14	D15
6	1	D16	D17	D18	D19	D20	D21	D22
7	D23	D24	D25	D26	D27	D28	D29	D30
8	1	D31	D32	D33	D34	D35	D36	D37
9	D38	D39	D40	D41	D42	D43	D44	D45
10	1	D46	D47	D48	D49	D50	D51	D52
11	D53	D54	D55	D56	D57	D58	D59	D60
12	1	D61	D62	D63	D64	D65	D66	D67
13	D68	D69	D70	D71	D72	D73	D74	D75
14	1	D76	D77	D78	D79	D80	D81	D82
15	D83	D84	D85	D86	D87	D88	D89	D90
16	1	D91	D92	D93	D94	D95	D96	D97
17	D98	D99	D100	D101	D102	D103	D104	D105
18	1	D106	D107	D108	D109	D110	D111	D112
19	D113	D114	D115	D116	D117	D118	D119	D120
20	1	D121	D122	D123	D124	D125	D126	D127
21	D128	D129	D130	D131	D132	D133	D134	D135
22	1	D136	D137	D138	D139	D140	D141	D142
23	D143	D144	D145	D146	D147	D148	D149	D150

Speech Frame Format continued

Octet no.	Bit number 1	2	3	4	5	6	7	8
24	1	D151	D152	D153	D154	D155	D156	D157
25	D158	D159	D160	D161	D162	D163	D164	D165
26	1	D166	D167	D168	D169	D170	D171	D172
27	D173	D174	D175	D176	D177	D178	D179	D180
28	1	D181	D182	D183	D184	D185	D186	D187
29	D188	D189	D190	D191	D192	D193	D194	D195
30	1	D196	D197	D198	D199	D200	D201	D202
31	D203	D204	D205	D206	D207	D208	D209	D210
32	1	D211	D212	D213	D214	D215	D216	D217
33	D218	D219	D220	D221	D222	D223	D224	D225
34	1	D226	D227	D228	D229	D230	D231	D232
35	D233	D234	D235	D236	D237	D238	D239	D240
36	1	D241	D242	D243	D244	D245	D246	D247
37	D248	D249	D250	D251	D252	D253	D254	D255
38	1	D256	D257	D258	D259	D260	C16	C17
39	C18	C19	C20	C21	T1	T2	T3	T4

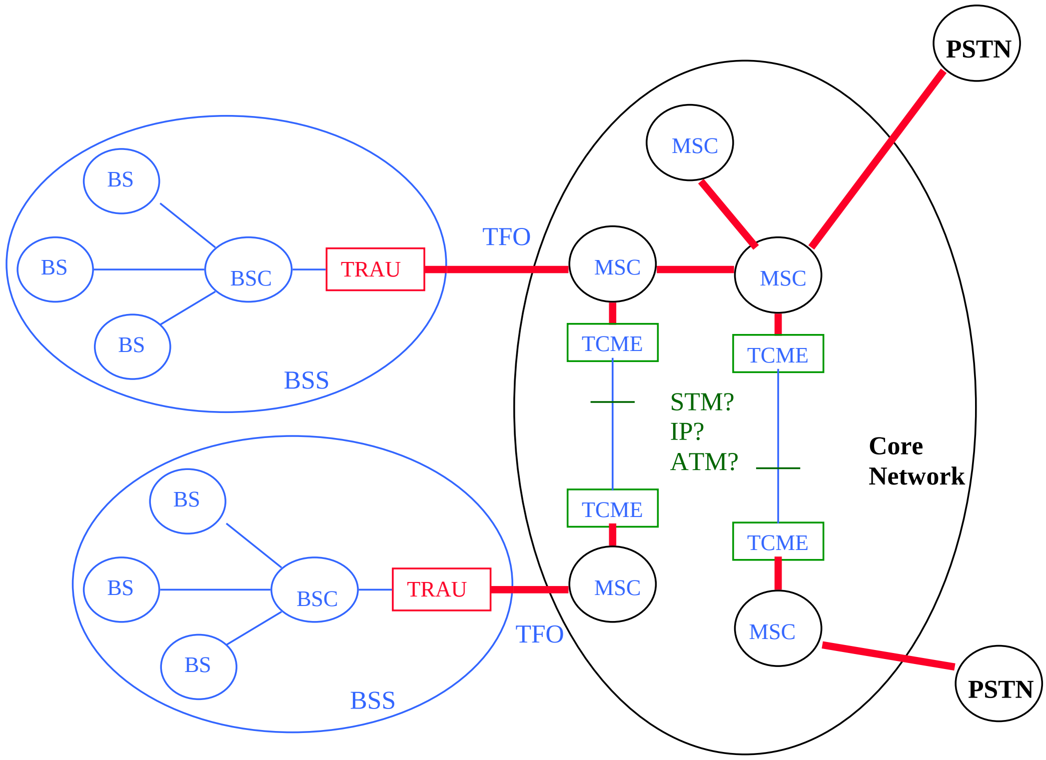
Bit rate 40 octets in 20 ms = 16.000 b/s → 1 frame lasts 20 ms

Speech rate 260 bits in 20 ms = 13.000 b/s (D1…D260)

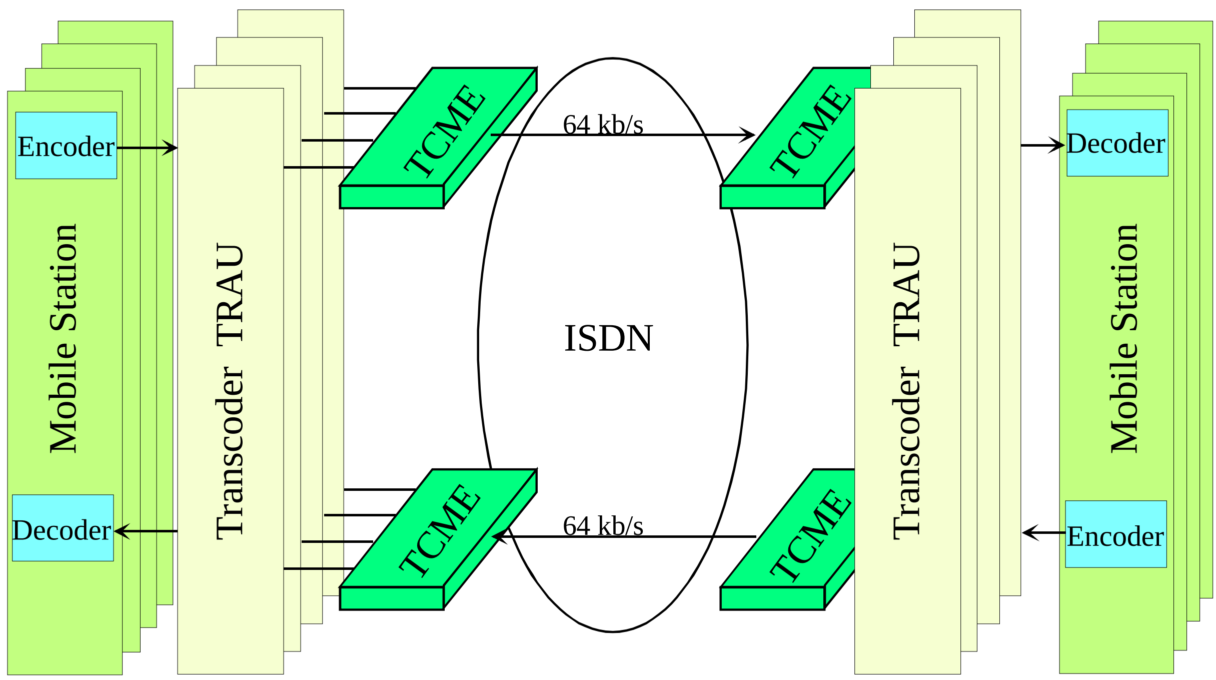
Control Information 21 bits in 20 ms = 1.050 b/s (C1…C21)

Bits C6 - C11: Control Bits for Time Alignment; T1…T4: Fill Bits for Time Alignment (may not be transmitted)

TFO in 2G Systems



Transmission Cost Reduction : TCME Architecture

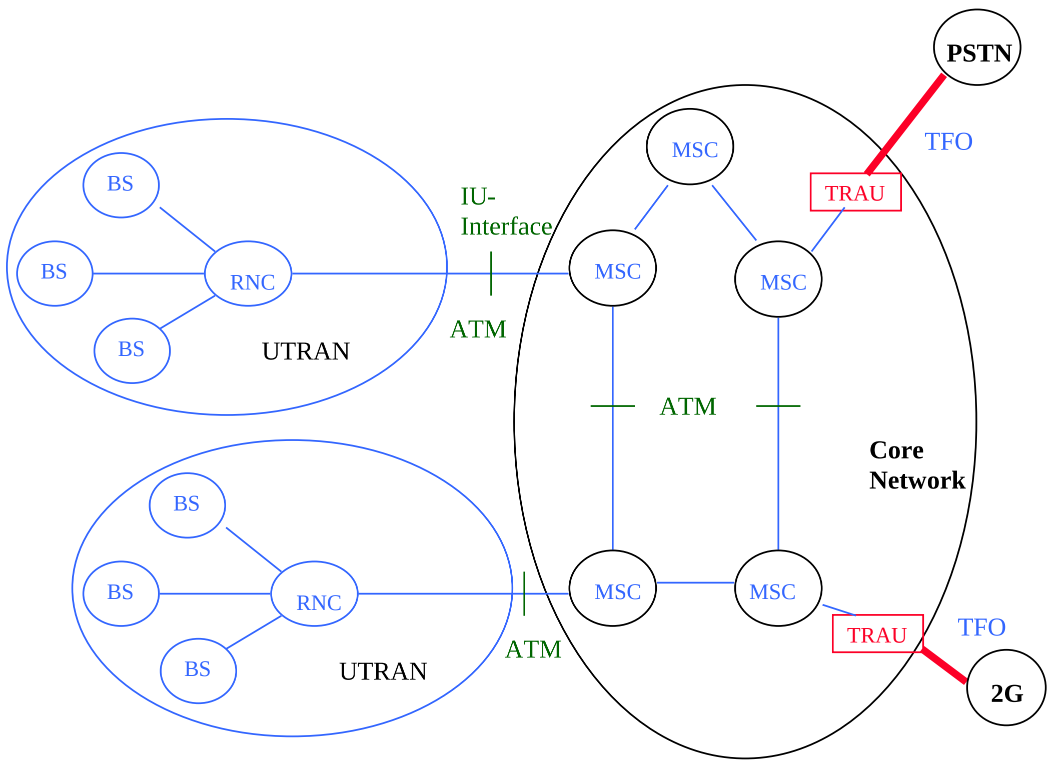


TCME: TFO specific Circuit Multiplication Equipment

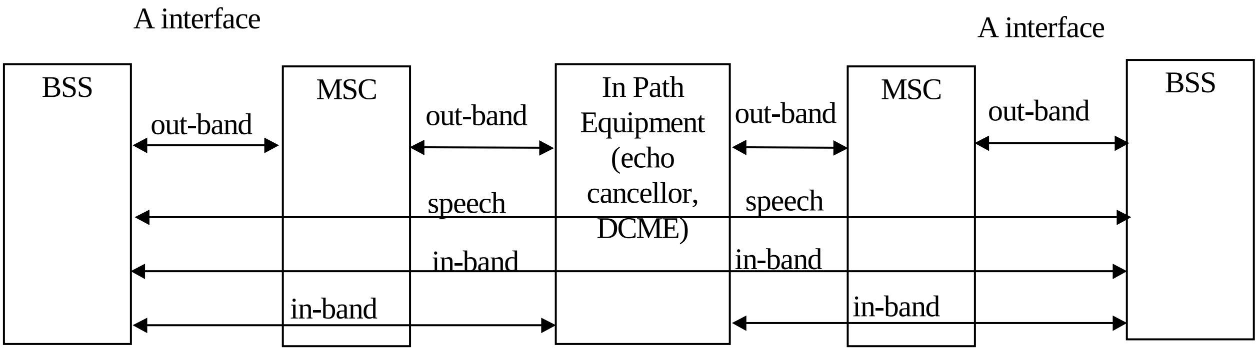
Tandem Free Operation | Transcoder Free Operation

TFO needs TRAU equipment	TrFO needs no transcoders
TFO is established by inband signalling	TrFO is established by out-of-band signalling
TFO can not change resources and Codec Type	TrFO can change resources and Codec Type
TFO works with minimal modifications	TrFO needs major protocol and network modification

TFO and TrFO in 3G Systems



TFO Reference Model



TFO:Tandem Free Operation
IPE: In Path Equipment