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VoLTE Optimization

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VoLTE Optimization principle and method

Introduction :

- VoLTE provides a first line telephony service with high voice quality and short call setup.
- Voice over LTE allows very fast call establishment (~1 sec) v/s CSFB towards 3G (~5 sec) and even more in case of CSFB towards GSM (~8 sec).
- IMS based voice standardized by 3GPP.

Domain selection :

- According to network registration system, domain selection is a process in which when UE originates or terminates the call terminal or network needs to select which network (2G, 3G, 4G) to be accessed.
- Terminating access domain selection (T-ADS) realizes the function of domain selection.

CSFB UE call procedure :

- CSFB UE-LTE access selected
- Detect available network
- Attach to the EPC & CS network over LTE
- Setup internet APN & do some browsing
- paging/call preparation between UE & MSC over LTE
- Place a call/ receive a call

VoLTE UE call procedure :

- VoLTE UE-LTE access selected
- Detect available network
- Attach to the LTE network
- Setup IMS APN & find P-CSCF(s)
- Register in IMS
- Place a call/receive a call (keeping current LTE access)

VoLTE Architecture

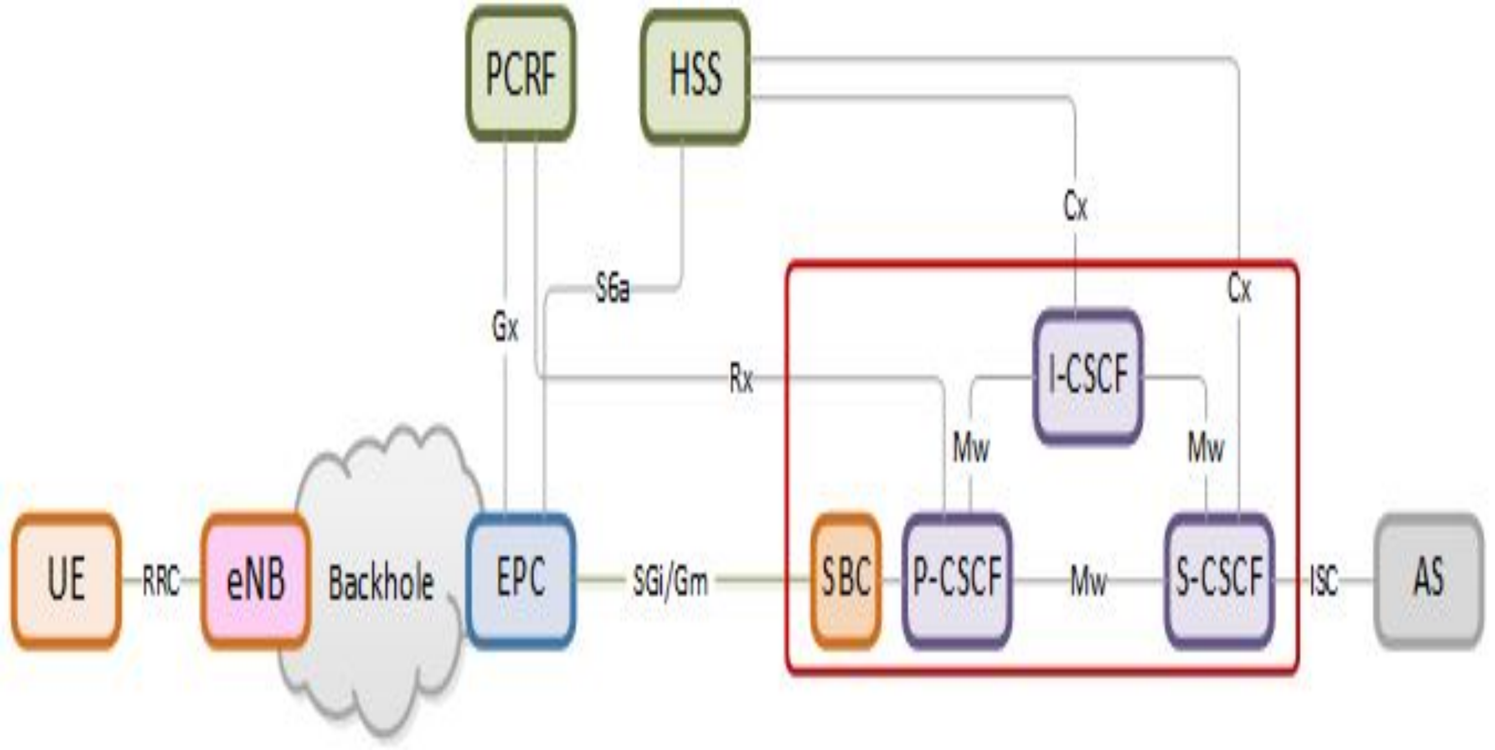
Architecture :

- VoLTE network architecture consists of E-UTRAN, LTE core , PDN and IMS.
- It interworks with 3G which consists of UTRAN, UMTS core and CS network.
- MME provides functions that allow LTE & 3G to interwork.

Architecture :

- Voice service is based on VOIP session, which is controlled by SIP session.
- Both devices need to be registered on IMS for VoLTE to take VoLTE calls.
- SIP proxy, VOIP application server (AS), media gateway (MGW) are co-located with MSC.

Architecture :



Architecture :

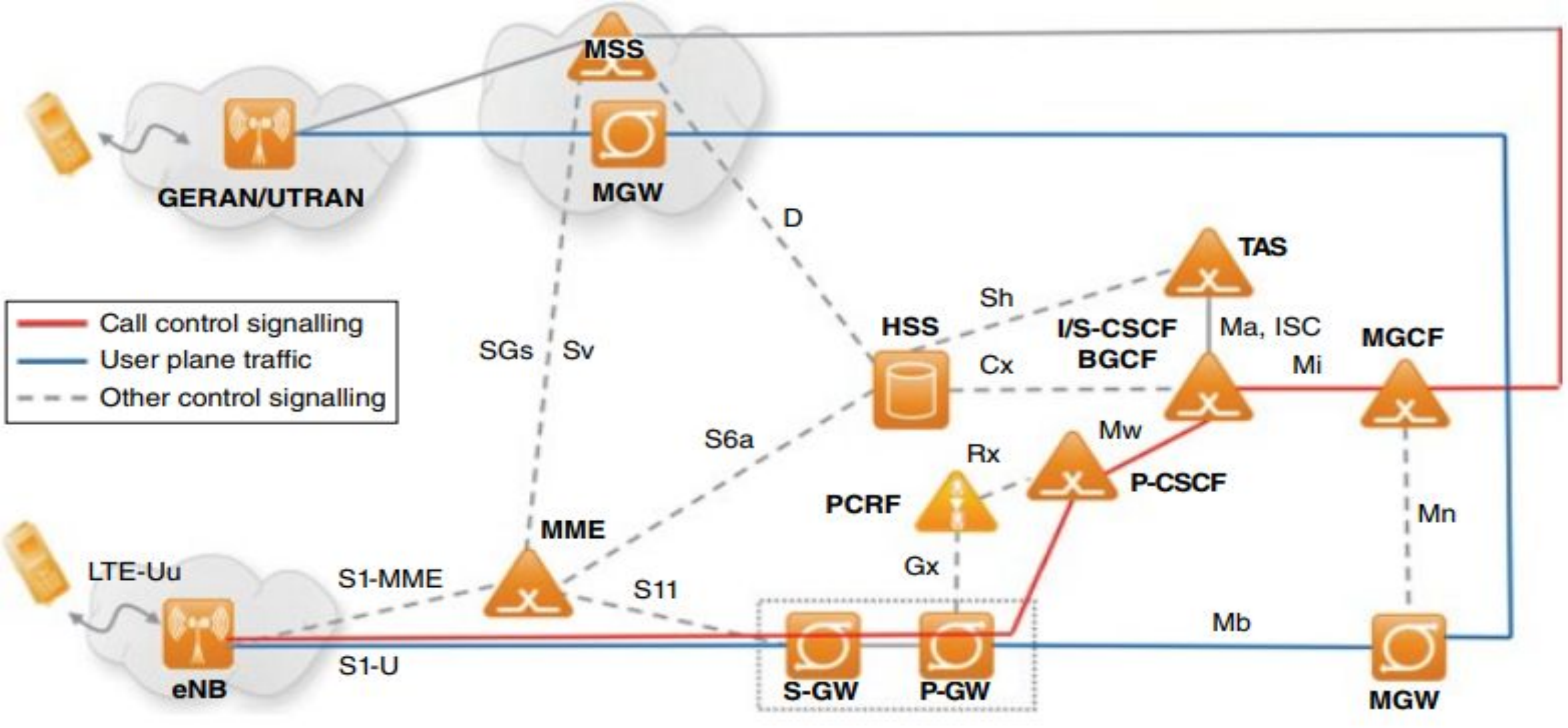
- IMS core network is also needed to be with centralized CSCF, HSS and VOIP AS , using dynamic policy provisioning with co-located PCRF.
- The P-CSCF forwards SIP messages received from the UE to an I-CSCF or S-CSCF and vice-versa .

VoLTE protocols

VoLTE protocols :

- SIP - session initiation protocol : it is a popular protocol used to create, modify and terminate multimedia sessions.
- SDP - session description protocol : it negotiates the multimedia characteristics of the session between sender and receiver.
- RTP - real time transport protocol : it adds a sequence number in order to identify lost packets.

VoLTE nodes & interfaces :



VoLTE interfaces & protocols :

Interfaces	Components LTE	Protocols
Sv	MSC server - MME (SGSN)	GTP-Cv2
12* Mj/Mg/Mx	MSS server - IMS-I-CSCF	SIP
Cx	HSS (NSN) - IMS core	diameter
Mw	P-CSCF - core IMS	SIP
Gm	P-CSCF - UE	SIP
Mb	ATGW - MSC server	RTP
Rx	AF - PCRF	diameter
Gx	PCEF - PCRF	diameter

VoLTE protocols :

- SSRC - synchronization source : all the packets have the same SSRC identifier indicating that they are from same source.
- CCRC - contribution source : it allows tracking of one or multiple sources for the packet.
- RTCP - real time transport control protocol : it monitors transmission statistics and quality of service information. For VoLTE, RTCP is not sent during active media transfer, but it is sent when the call is on hold.

SIP messages code :

100 Trying
180 Ringing
181 Call Is Being Forwarded
182 Queued
183 Session Progress

200 OK
202 Delivered

300 Multiple Choices
301 Moved Permanently
302 Moved Temporarily
305 Use Proxy
380 Alternative Service

400 Bad Request
401 Unauthorized
402 Payment Required
403 Forbidden
404 Not Found
405 Method Not Allowed
406 Not Acceptable
407 Proxy Authentication Required
408 Request Timeout
409 Conflict
410 Gone
411 Length Required
413 Request Entity Too Large
414 Request-URI Too Long
415 Unsupported Media Type
416 Unsupported URI Scheme

420 Bad Extension
421 Extension Required
423 Registration Too Brief
480 Temporarily Unavailable
481 Call/Transaction does not exist
482 Loop Detected
483 Too Many Hops
484 Address Incomplete
485 Ambiguous
486 Busy Here
487 Request Terminated
488 Not Acceptable Here
491 Request Pending
493 Undecipherable

500 Server Internal Error
501 Not Implemented
502 Bad Gateway
503 Service Unavailable
504 Gateway Time-out
505 Version Not Supported
513 Message Too Large
600 Busy Everywhere
603 Decline
604 Does Not Exist Anywhere
606 Not Acceptable

VoLTE technical summary

VoLTE technical summary :

- VoLTE bearer management, includes PDN connection for IMS APN, signalling bearer setup, P-CSCF discovery, home routed PDN connection/APN for Ut, handling of loss of PDN connection, signalling & GBR bearer.
- IMS feature part, includes ISIM based authentication.
- IMS media, includes AMR narrowband and wideband codec and payload format.
- SMS includes SMS over IMS and SMSoSGs.

VoLTE capability in UE :

- RRC UE capability information message.
- The FGI (feature group indicator) indicates functionalities supported by UE.

VOIP/video QoS and features :

- Voice is sensitive to data loss , but robust coding and well - functioning error concealment unit makes voice more tolerant to data loss than other media types.
- Video telephony is also sensitive to data loss, video telephony requirements on frame errors are almost a factor of 10 less than for voice.

VOIP/video QoS and features :

- TCP and SIP retransmissions are the mechanisms that guarantee that the data transfer becomes error free even though the EPS bearer introduces packet loss.
- PCC (policy and charging control) enables QoS supervision and control for the media parts of the SIP session.
- PCRF (policy and charging rule function) supports PCC procedures and makes policy and charging decisions based on input from user subscription information , services info and so on.

Voice codec :

- Possible codecs for voice : EVCR-A , EVCR-B, AMR, AMR-WB (wide-band) and EVS.
- Tandem free operation (TFO) and transcoder free operation (TrFO) must be supported in the IMS core for CS interworking.
- Speech latency requirement : no more than 150 ms is preferred.
- Packet loss : 1%

Video codec :

- 3GPP has standardized that QCI1 is used for the transmission of voice RTP packets and & QCI2 is used for transmission of video RTP packets.
- QCI1 & QCI2 will be triggered from PCRF on demand.
- The recommended video codec for conversational video service in VoLTE is H.264.

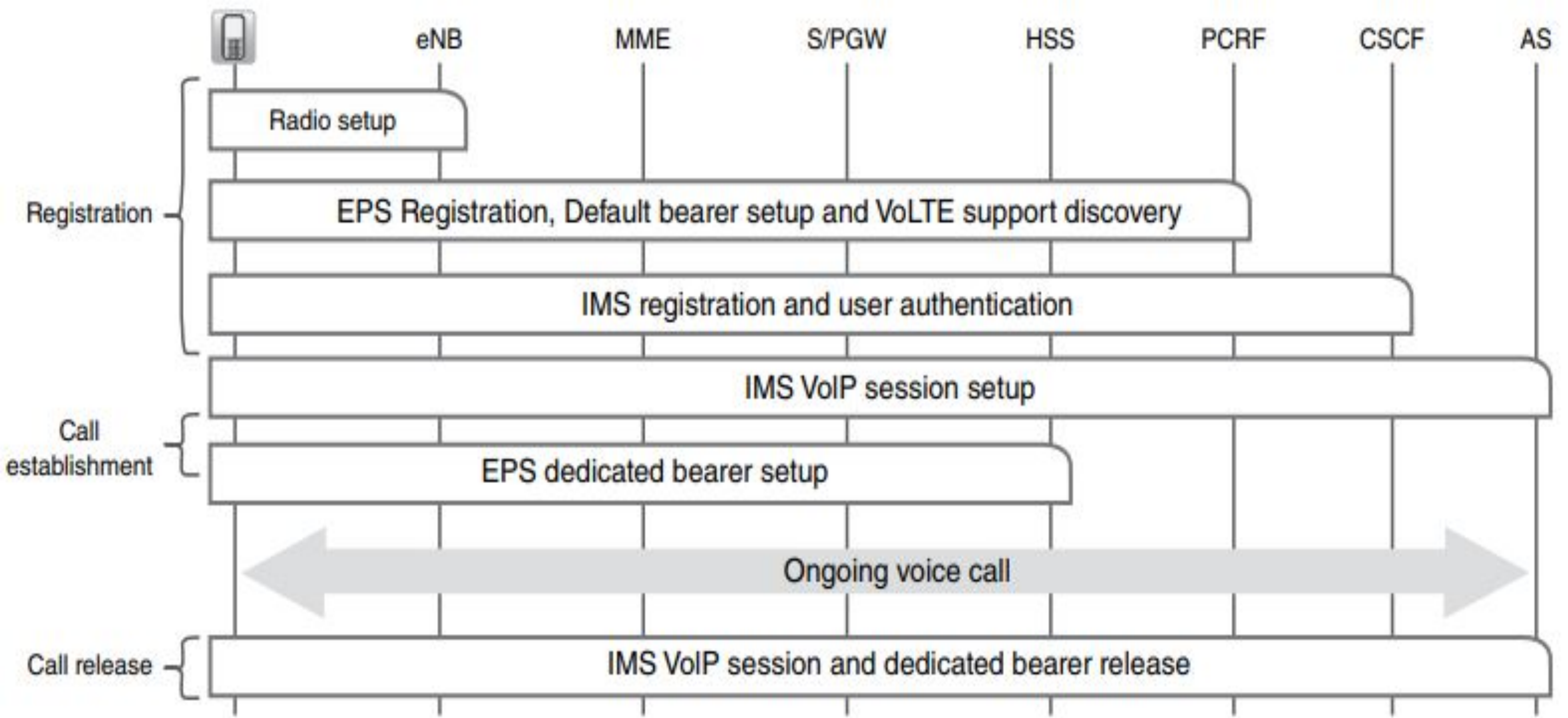
Radio bearer for VoLTE :

- For VoLTE, PDN connection handles flow of IP packets , that are labelled with UE IP addresses between UE and PDN.
- It is represented by UE IP address & APN.

RLC UM :

- An RLC entity can be configured to operate in T M, U M or AM mode.
- RLC UM provides a unidirectional data transfer service without sending any feedback to the transmitting entity.

High level E2E VoLTE call flow :



VoLTE capacity & coverage

VoLTE capacity :

- VoLTE capacity mainly depends on scheduling capacity and PUSCH capacity.
- VoLTE capacity is defined according to the number of QCI 1 bearers managed by the cell.

VoLTE coverage :

- VoLTE deployments have targeted use of HD voice codecs such as AMR12.65 Kbps for improved voice quality.
- VoLTE has additional IP, PDCP, RLC, and MAC header overhead compared to circuit switched voice.
- With segmentation, VOIP packets are split into smaller parts in order to distribute transmission over several TTI's.

RLC segmentation & TTI bundling

RLC Segmentation :

- When a VoLTE UE moves towards the cell edge, SINR received by eNB starts to decrease due to UE power limitation.
- Packet segmentation algorithm means segmenting the VOIP packets from PDCP into multiple smaller packets.
- This algorithm is not an event triggered mechanism , it is done automatically and only for UE's with poor radio channel.

RLC Segmentation :

- It reduces the payload bit per transmission and increases the amount of power per transmitted bit.
- Link adaptation in dynamic schedular reduces MCS level to a point where eventually a VOIP packet is segmented into many separate smaller MAC SDU’s and improves the link budget.

TTI Bundling :

- TTI bundling is intended particularly for addressing LTE uplink budget issues and balance the UL coverage with the DL coverage footprint.
- It can be very beneficial when the UE is close to the cell edge specifically for VoLTE services.
- When TTI is activated , TTI bundling uses four automatic retransmissions in four consecutive uplink TTI with a common ACK/NACK for HARQ.

VOIP packet size with RLC segmentation & overhead analysis :

	no segment			2 segments			4 segments			8 segments		
	Segment Size	L1 SDU		Segment Size	L1 SDU		Segment Size	L1 SDU		Segment Size	L1 SDU	
Voice Codec	[bits]	[bits]	Segment overhead	[bits]	[bits]	Segment overhead	[bits]	[bits]	Segment overhead	[bits]	[bits]	Segment overhead
AMR-NB 1.8	120	120	0%	80	160	33%	64	256	100%	56	448	233%
AMR-NB 4.75	176	176	0%	112	224	23%	80	320	68%	64	512	159%
AMR-NB 5.15	184	184	0%	112	224	22%	80	320	65%	64	512	152%
AMR-NB 5.9	200	200	0%	120	240	20%	80	320	60%	64	512	140%
AMR-NB 6.7	216	216	0%	128	256	19%	88	352	56%	64	512	130%
AMR-NB 7.4	232	232	0%	136	272	17%	88	352	52%	64	512	121%
AMR-NB 7.95	240	240	0%	144	288	17%	96	384	50%	72	576	117%
AMR-NB 10.2	288	288	0%	168	336	14%	104	416	42%	72	576	97%
AMR-NB 12.2	328	328	0%	184	368	12%	112	448	37%	80	640	85%
AMR-WB 1.75	120	120	0%	80	160	33%	64	256	100%	56	448	233%
AMR-WB 6.6	216	216	0%	128	256	19%	88	352	56%	64	512	130%
AMR-WB 8.85	264	264	0%	152	304	15%	96	384	45%	72	576	106%
AMR-WB 12.65	336	336	0%	192	384	12%	120	480	36%	80	640	83%
AMR-WB 14.25	368	368	0%	208	416	11%	128	512	33%	88	704	76%
AMR-WB 15.85	400	400	0%	224	448	10%	136	544	30%	88	704	70%
AMR-WB 18.25	448	448	0%	248	496	9%	144	576	27%	96	768	63%
AMR-WB 19.85	480	480	0%	264	528	8%	152	608	25%	96	768	58%
AMR-WB 23.85	560	560	0%	304	608	7%	176	704	21%	112	896	50%

TTI Bundling :

- TTI bundling is intended particularly for addressing LTE uplink budget issues and balance the UL coverage with the DL coverage footprint.
- It can be very beneficial when the UE is close to the cell edge specifically for VoLTE services.
- When TTI is activated , TTI bundling uses four automatic retransmissions in four consecutive uplink TTI with a common ACK/NACK for HARQ.

TTI bundling optimization

TTI Bundling optimization :

- TTI bundling will take four subframes of uplink, which impact the capacity greatly, for TTI bundling optimization , it's reasonable and necessary to restrict the TTI bundling user number, based on configured SINR and TTI bundling number.
- TTI bundling increases the probability of successful transport block decoding by eNB.

TTI Bundling optimization :

- The eNB periodically monitors every 100 ms the eligible UE's for TTI bundling activation/deactivation.
- The eNB periodically monitors every 100 ms the eligible UE's for TTI bundling activation/de-activation.
- Once the criteria for entering TTI bundling mode are fulfilled, eNB triggers intra-cell handover procedure by sending RRC connection re-configuration message toward UE.

TTI Bundling optimization :

- TTI bundling activation criteria include load criteria and poor RF performance conditions.
- For load criteria, current total GBR PRB utilization in the cell should be less than TTI bundling activation load threshold, number of users in the cell already in TTI bundling configuration should be less than the maximum number of TTI bundling users.

TTI Bundling gain :

TTIB	UE#	MoS	UL PRB	MCS	Tx Power PUSCH	PUSCH BLER[%]	Phy Thrp [kbps]	PDCP Thrp [kbps]	DL RSRP [dBm]
OFF	1	2.4	5.0	0.1	24.0	54.4	47	13	–118.7
	2	1.8	4.9	0.1	23.9	39.6	38	15	–120.6
ON	1	3.2	3.0	10.1	23.5	0.6	125	14	–122.1
	2	3.0	3.0	12.0	23.3	0.1	128	15	–115.6

Coverage gain with RLC segmentation & TTI Bundling :

- Packet segmentation algorithm is used as an extension to link adaptation for UL coverage improvement, it is done automatically and only for UE’s with poor radio channel.
- The UL power control function manages to keep the UE received power to a constant value.

MCS/TBS/PRB selection :

- Too much segmentation can lead to excessive queuing delay since each segment of VOIP packets needs separate grant and uses separate HARQ process and PDCCH consumption is higher.
- The TBS is obtained by adding the PDCP/RLC/MAC overhead
- TBS = VOIP codec VOIP frame size + 4 bytes (RoHC header) + MAC, RLC overhead.

TBS index table for VOIP :

	NPRB									
I_{TBS}	1	2	3	4	5	6	7	8	9	10
0	16	32	56	88	120	152	176	208	224	256
1	24	56	88	144	176	208	224	256	328	344
2	32	72	144	176	208	256	296	328	376	424
3	40	104	176	208	256	328	392	440	504	568
4	56	120	208	256	328	408	488	552	632	696
5	72	144	224	328	424	504	600	680	776	872
6	328	176	256	392	504	600	712	808	936	1032
7	104	224	328	472	584	712	840	968	1096	1224
8	120	256	392	536	680	808	968	1096	1256	1384
9	136	296	456	616	776	936	1096	1256	1416	1544
10	144	328	504	680	872	1032	1224	1384	1544	1736
11	176	376	584	776	1000	1192	1384	1608	1800	2024
12	208	440	680	904	1128	1352	1608	1800	2024	2280
13	224	488	744	1000	1256	1544	1800	2024	2280	2536
14	256	552	840	1128	1416	1736	1992	2280	2600	2856
15	280	600	904	1224	1544	1800	2152	2472	2728	3112

TBS index table for VOIP :

16	328	632	968	1288	1608	1928	2280	2600	2984	3240
17	336	696	1064	1416	1800	2152	2536	2856	3240	3624
18	376	776	1160	1544	1992	2344	2792	3112	3624	4008
19	408	840	1288	1736	2152	2600	2984	3496	3880	4264
20	440	904	1384	1864	2344	2792	3240	3752	4136	4584

MCS/TBS/PRB selection :

- For optimization, MCS #PRB’s is one of the items which need to consider.
- Required TBS, combination of MCS #PRB’s and SINR requirements in VOIP packet transmission.
- After enforcing the minimum number of PRB’s , MCS override approach is:
  2 PRB grant→ min MCS= A
  3 PRB grant→ min MCS= B
  4 PRB grant→ min MCS= C

Link budget

Link budget :

- Lowest bit rate providing excellent speech quality in a clean environment is 12.65 kbps.
- Higher bit rates are useful in background noise conditions and for music.
- The different voice transport formats (modulation, coding schemes, no. of resource blocks and transport block size) have different SINR requirements.

Link budget :

- The UL link budget, maximum acceptable UL path loss (lpmax)

Lpmax=P ue,rb - S eNB - B iul - B lnf - L bl - L bpl + Ga - Lj

SINR requirements for voice codec :

TBS	MCS	TBS	Modulation	Veh-A, 3 km/hr	Veh-A, 50 km/hr	Veh-A, 120 km/hr	Ped-A, 3 km/h
AMR 12.2, 328	6	6	QPSK	–5.6/1.4	–4.8/2.7	–3.1/3.2	–5.6/0.4
AMR 4.5, 144	10	10	QPSK	–8.0/–2.9	–7.4/–1.9	–5.8/–1.3	–7.9/–3.7
AMR 5.9, 224	14	13	16QAM	–1.5/1.7	–1.1/2.9	0.73/3.4	–1.5/0.6
AMR 7.95, 256	15	14	16QAM	–0.9/3.0	–0.06/3.8	1.5/4.4	–0.9/1.2
AMR 12.2, 328	17	16	16QAM	–0.5/0.9	0.3/1.5	1.9/2.0	–0.5/–1.1

VoLTE link budget under TDD 20 MHz :

Uplink	AMR12.2 k	AMR23.85 k	PUSCH	
Antenna configuration	1Tx*8Rx	1Tx*8Rx	1Tx*8Rx	1Tx*8Rx
Modulation	QPSK	QPSK	QPSK	QPSK
Code rate	0.16	0.2	0.31	0.31
User data rate(kbps)	104	164	256	512
# of Occupied RBs	3	3	8	16
UE EIRP				
MS power (dBm)	23	23	23	23
MS EIRP (dBm)	18.2	18.2	14.0	11.0
eNB sensitivity				
BS antenna gain (dBi)	16.5	16.5	16.5	16.5
BS diversity gain (dB)	*0*	*0*	*0*	*0*
Transmission line loss (dB)	*0.50*	*0.50*	*0.50*	*0.50*
Thermal noise (kT) (dBm/Hz)	–174.0	–174.0	–174.0	–174.0
Subcarrier separation (KHz)	15	15	15	15
BS noise power (dB)	–117.9	–117.9	–117.9	–117.9
BS noise figure (dB)	3.5	3.5	3.5	3.5

VoLTE link budget under TDD 20 MHz :

SNR for MCS Level - 1%BLER (dB)	–5.3	–4.3		
SNR for MCS Level - 10%BLER(dB)			–7.8	–7.8
BS sensitivity - composite	–123.2	–122.2	–125.7	–125.7
Shadow fading margin (dB)	8.3	8.3	8.3	8.3
Interference margin (dB)	4.0	4.0	4.0	4.0
Penetration loss (dB)	20	20	20	20
MS body loss (dB)	3	3	0	0
Terminal loss (dB)	6	6	6	6
Total system margin (dB)	41.3	41.3	38.3	38.3
Outdoor maximum allowable path loss	136.6	135.6	137.9	134.9

VoLTE delay

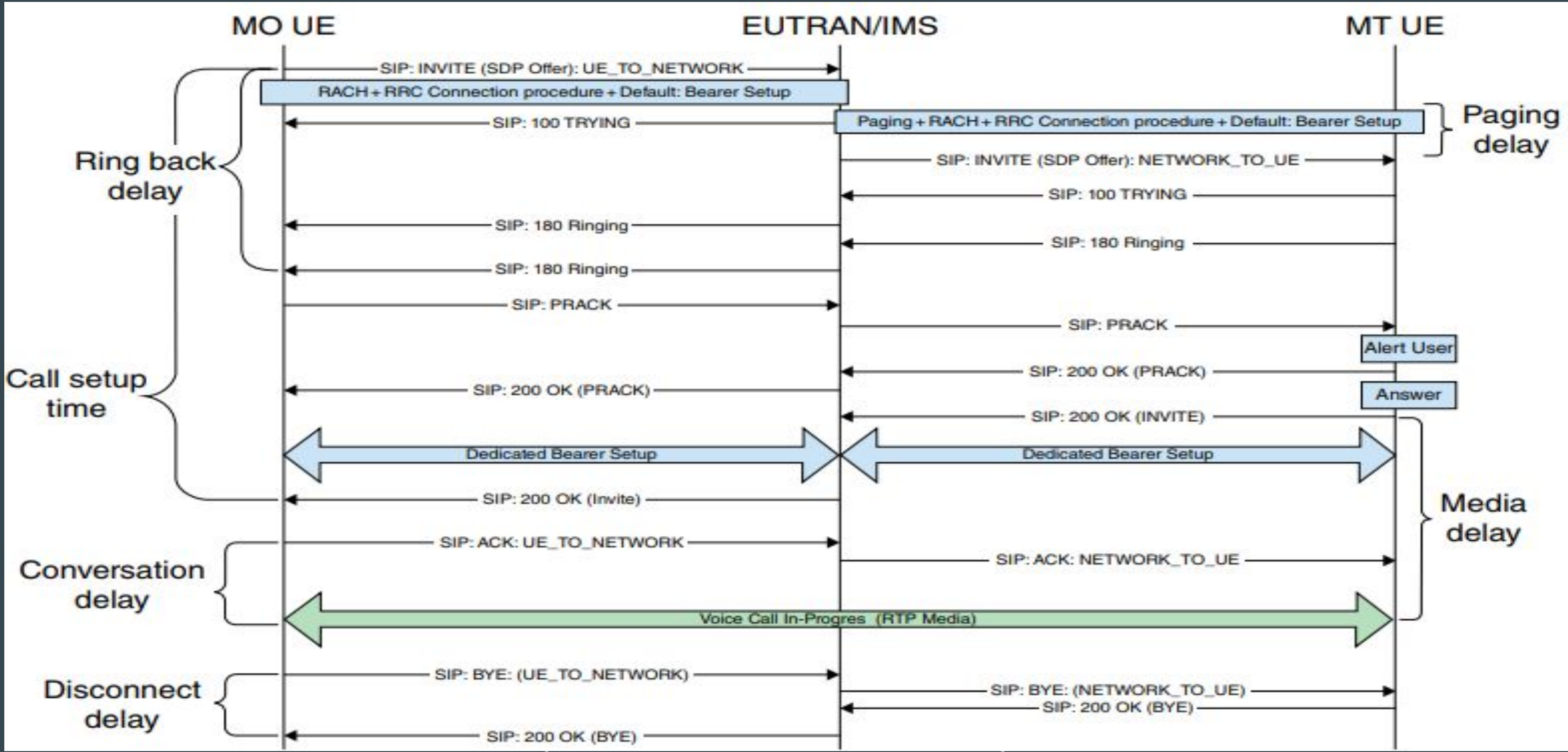
VoLTE Delay :

- Delay is crucial for VOIP, delay target must be controlled by the schedular.
- The selection of users for scheduling in given TTI is based on
  C(t) = P F(t) * DT(t)
  PF(t) is proportional fair term
  DT(t) is delay term
  Delay term is based on VOIP packet delay

Comparison of VoLTE PUSCH & UL control channel :

Coverage comparison			DU	U	SU	RU	
RACH	Msg1	Format0	1.4 km	2.3 km	4.5 km	16.8 km	
		Format2	1.7 km	2.7 km	5.2 km	19.9 km	
		Format3	1.7 km	2.7 km	5.3 km	20.0 km	
	Msg3 (HARQ = 3)	56 bits	1.1 km	1.7 km	3.6 km	13.6 km	
		104 bits	**1.0 km**	**1.5 km**	**3.2 km**	**12.2 km**	Limitation
	Msg3 (HARQ = 5)	56 bits	1.3 km	2.0 km	4.3 km	16.4 km	
		104 bits	1.1 km	1.8 km	3.9 km	14.7 km	
WB-SRS	WB-SRS	48 RBs	1.7 km	2.8 km	6.4 km	24.3 km	
VoLTE WB-AMR 12.65	PUSCH	SPS and No TTI-B	0.7 km	1.1 km	2.5 km	7.7 km	
		Dynamic scheduling	0.9 km	1.4 km	3.0 km	10.2 km	
		max 4 segments	1.0 km	1.6 km	3.6 km	13.4 km	
		TTI-B	1.2 km	1.9 km	4.2 km	14.3 km	

Typical VoLTE delay :



Impact on device & network :

- Option #2 requires deployment of 5GC and update of NR gNB to support both NSA and SA option in parallel.
- Option #2 has impact on E-UTRAN connected to EPC to support inter-RAT mobility , IMS to support 5GS QoS management and also on UE.

Call setup delay

Call setup time :

- Call setup time is measured entirely from the UE, which originates the call.
- Typical call setup time is 3 to 5 seconds.
- In a live network, engineers usually modify call setup time calculation formula with appropriate end trigger depending on call case.

Combinations of call cases :

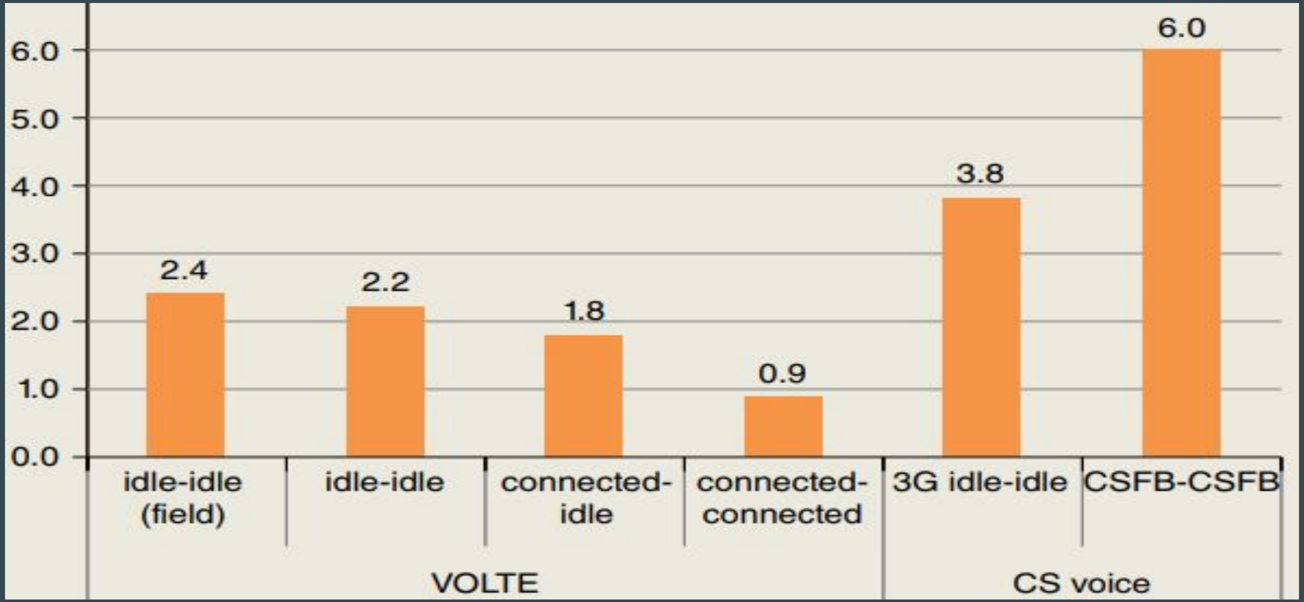
- No preconditions and no early media - end trigger is 180 rings.
- Preconditions and no early media - end trigger is 180 rings.
- No preconditions and early media - end trigger is 200 OK (PRACK).
- Preconditions and early media - end trigger is 200 OK (UPDATE).

Reasons for long call setup time :

- RACH procedure
- Paging strategy
- Radio condition affecting dedicated bearer setup time.
- Rx/Gx interface issues
- Retransmission of signaling messages
- Traffic congestion/overload

Conversation start delay :

- It is the time from the instant the call has been answered until the media RTP stream starts in either direction.



RTP delay

RTP delay :

- It represents the one way IP transport delay between 2 UE's including both the network delay and also the LTE and IP stack processing in the UEs.
- To avoid time synchronization issues, the RTP delay is measured one way in both directions between the UE's involved in the test call.

RTP delay :

- The total delay is around 180 ms + backhaul delay + IP network delay.
- Also the schedular on the eNB will contribute some delay since a VoLTE packet is scheduled every 40 ms (20ms).
- Importance for VoLTE is the ability to achieve end - to - end latency delay for voice packets such that delay meets user expectations.

Delay budget of network entities :

Delay Component	Max value (ms)	Range (ms)	Remark/Assumptions
UE delay UL/DL	44/39	31–44	15 ms processing time is assumed as worst case number for UE implementation
Air interface one-way delay (UL/DL)	48/41	4–48	For 6 HARQ transmissions using dynamic scheduling as worst case
eNB delay	8	2–8	eNB processing delay for scheduling and packet L1/L2 processing
SGW	0.5	0.1–0.5	Packet forwarding
PDN-GW	0.5	0.1–0.5	Packet forwarding
S1-U or S5	–	2–15	Propagation delay is mainly proportional to distance (5us/km)
IP Network	–	18–41	Assuming 2 ms processing and queuing, and 2000 miles OC3

Handover delay and optimization

Handover delay :

- Voice interrupt time during handover means discontinuity of voice media flow due to HO in UL & DL directions respectively.
- Uplink voice interruption is measured in the remote UE.
- The feature of data forwarding at intra-LTE HO will minimize the voice interruption and DL packet loss at X2 HO.

Handover delay :

- For data forwarding during X2 and S1 handover, only fresh S1 SDU's from old S1 path will be forward since VoLTE bears utilize UM RLC which is not lossless.
- If in case, data forwarding is not possible, it leads to throughput degradation.
- The VoLTE characteristics requirement specifications require a HO performance of less than 50 ms speech interruption time in 90% of the cases.

Handover optimization :

- SIB read failure during HO
- Total HO interrupt time is 781 ms due to a RLF happened during HO.

Intra LTE HO & eSRVCC

Intra LTE HO

Intra frequency HO :

- To optimize mobility related parameters , HO success rate is used as a criteria.
- Voice quality related criteria also required to optimize HO related parameters.
- If handovers are triggered too late, high DL BLER will cause HO command decode failures.

Intra frequency HO :

- RTP loss primarily occurs at HO regions due to delayed HO's and poor UL/DL conditions.
- Reducing the time duration of declaring RLF can also reduce RTP loss during HO failures as VoLTE is more time sensitive than data traffic.

Inter frequency HO :

- inter -frequency HO's are measurement based and prepared and executed after a A 5 MR is received at the eNB.
- Measurement based inter-frequency HO with A5 measurement in the UE occurs after receiving an A2 (bad coverage) MR.

Single radio voice call continuity procedure

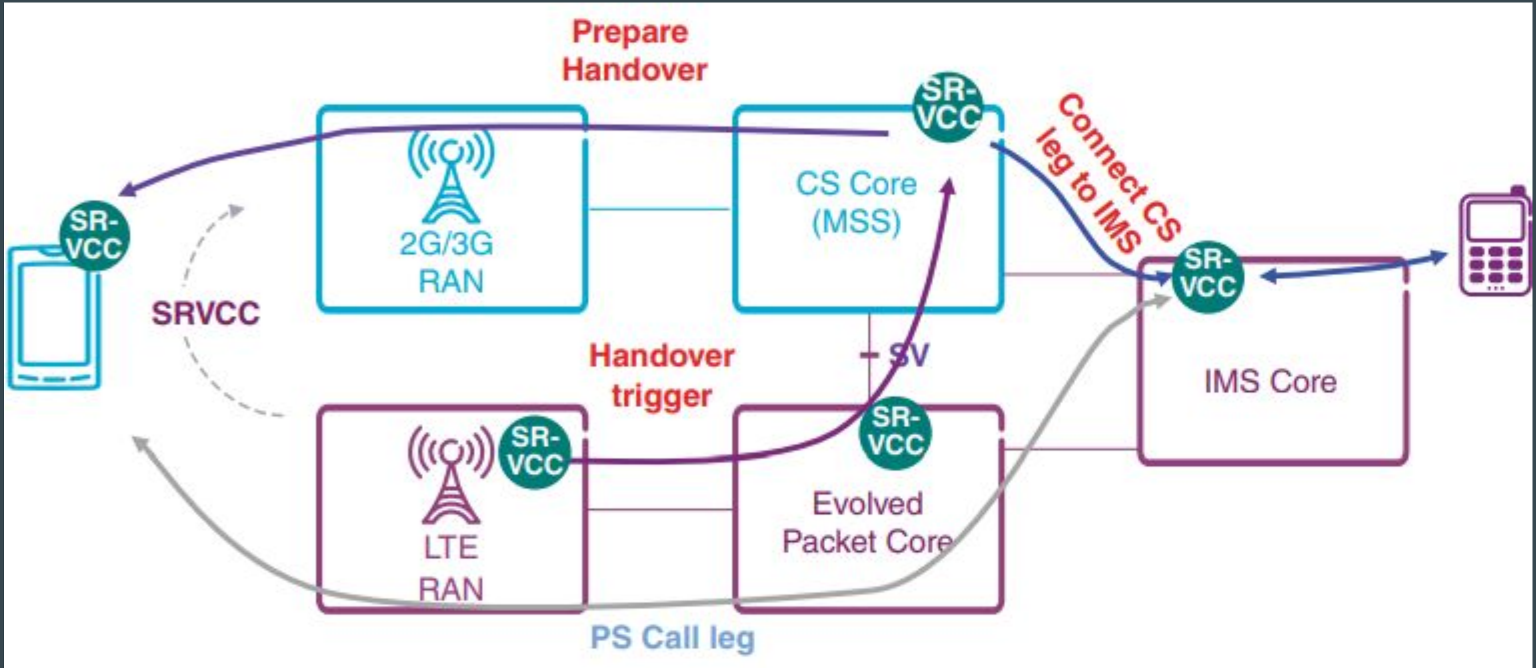
SRVCC :

- For SRVCC, single radio means that the UE transmits and receives signals on only one radio access at a given time (LTE, WCDMA or GSM) in order to minimize power consumption and radio emission.
- SRVCC to 2/3 G feature provides voice call continuity from a PS access domain to a CS access that are anchored in IMS.

Functionality :

- During IRAT handover, there are two call legs that the MSC must initiate, one towards the new radio link and the other one to the IMS where the on going call must be connected again.

Functionality :



UE SRVCC capability :

There are four kinds of method for eNB to get UE's SRVCC capability:

- S1 initial context setup request
- RRC UE capability information
- X2 HO request
- S1 HO request

SRVCC parameters :

During SRVCC, two parameters are used :

- C-MSISDN (correlation MSISDN)
- STN-SR for SRVCC

SRVCC parameters optimization

Handover parameters :

- If SRVCC handovers are supported and allowed by both UE and the network, it is stored in both HSS and MME.
- VoLTE calls, which are still in pre (alerting) phase and will enter bad LTE coverage, will be dropped if the SRVCC handover procedure is initiated.
- The threshold values used to configure UE for event A1,A2,A4,A5 , B1 and B2 measurements should be adjusted by VOIP specific offset parameters.

SRVCC related timer :

- The timers are related to SRVCC preparation phase and SRVCC execution phase.
- During the SRVCC preparation phase, the related timer is : GSM/WCDMA TS1 reloc prep , GSM/WCDMA TS1 relocoverall

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aSRVCC and bSRVCC :

- VoLTE performance needs to be good so SRVCC threshold is typically set to a relatively higher threshold (e.g -116 dbm).
- In case the function of a/b SRVCC is not enabled , the UE cannot reselect to 3G and remains in LTE, but cannot make any LTE call.
- Operator has to avoid SRVCC occuring during ringing as much as possible.

SRVCC Failure

SRVCC failure :

Outgoing SRVCC failures can be spotted by :

- S1AP UE context release command message received from MME with cause other than “successful HO”.
- S1AP reset received from the MME or S1AP reset is eNB initiated.

SRVCC failures causes :

- HO preparation failure.
- TS1 relocprep for SRVCC HO to 2/3G timer expiry.
- RRC connection re-establishment on serving/other cell requested by UE.
- TS1 relocoverall for SRVCC HO to 2/3G timer expiry.

Network quality and subjective speech quality

Network quality & subjective speech quality :

- VoLTE average call holding time is 90 seconds, but call will reach just upto 24kbps.
- In VoLTE, digitized voice must share network resources with a variety of other bandwidth hungry services.
- Continuous coverage, optimized mobility , balanced DL and UL , reduced overlap and interference is desired in VoLTE performance.

Network quality & subjective speech quality :

- VoLTE speech quality consists of subjective and objective speech quality estimations, speech path delay (SPD) and frame error rate (FER) measurements.

VoLTE experience KPI's :

No.	KPI name	Source	Target values
1	MOS: VoLTE to VoLTE WB12.65/12.2NB	Drive test	>3
2	Speech path delay	Drive test	<200 ms
3	Call setup time (ECM connected mode)	Drive test	<4 Sec
4	Handover interruption time (CP + UP)	Drive test	<200 ms
5	Jitter of Rx RTP packet	Drive test	<60 ms
6	IMS registration	Drive test	<2 Sec
7	Packet loss rate	Drive test	<1 %
8	SRVCC Interruption time	Drive test	<300 ms
9	Drop call rate	OSS Stats	<1%
10	Access failure	OSS Stats	<1 %
11	SRVCC success rate	OSS Stats	>98.5%
12	Handover success rate	OSS Stats	>99%
13	UL/DL BLER	OSS Stats	<10 %
14	Paging success rate	OSS Stats	>99%

Factors affecting performance of VoLTE :

- Jitter buffer management
- Noise reduction
- Echo canceller
- Speech level
- Audio fidelity
- Audio level compensation
- Speech codec
- HO interruption time
- Delay
- Frame loss

Bearer latency :

- To maintain the voice quality of wireless VOIP network, there is a need to maintain mouth to ear user plane delay to be less than 200 msec.
- To measure voice satisfaction, the end to end delay is also important.
- There are 5 major call cases in VoLTE calls, in which end to end delays are different.

Mean opinion score

Voice quality :

- Speech quality is one of the most important factors for the attractiveness of a speech service.
- Voice quality in VoLTE is measured using MoS score.
- VoLTE speech quality consists of subjective and objective speech quality estimations , speech codec, discontinuous transmissions, network echo canceller, speech path delay and FER measurements.

Video quality :

- Video quality metrics for VoLTE video is PEVQ.
- A pre-defined reference file with video samples is injected into the originating UE, traversing over LTE/EPC and received in the terminating UE.

Jitter :

- The variance in the inter-arrival time of the RTP packets.
- It is the difference between the time interval between the arrival of two successive packets.
- Normally jitter buffer typical size range is 40 ms to 80ms , thus jitter of about 40 ms between device and eNB is acceptable.

Packet loss :

- The number of RTP packets not received by the UE is calculated based on the RTP timestamp , sequence number and SSRC.
- The number of dropped packets E is calculated per RTP flow by adding up the number of RTP packets lost.

One way audio :

- One way audio occurs when RTP flows in only one direction.
- One way audio is also caused when a call is abruptly released by IMS due to an RTP time out.
- When one way audio happened right after call establishment, the causes of the problem could also be firewall issue or codec issue.
- When one way audio happened during ongoing call, especially during X2 handovers , the RTP UL/DL paths have to be transferred from source eNB to target eNB.

PDCP discard timer operation :

- During HO, delivery of PDCP PDU’s to lower layers is stopped and they are buffered within the UE PDCP buffer.
- Due to poor RF condition , only small grants are allocated, it takes quite a long time to transmit one single PDU.

Optimization

Optimization :

- VoLTE performance can be derived by collecting drive data, MoS scores, call traces and network statistics.
- To identify whether a particular VoLTE session is experiencing any problem, that session needs to be characterized with indicators at all possible interfaces.

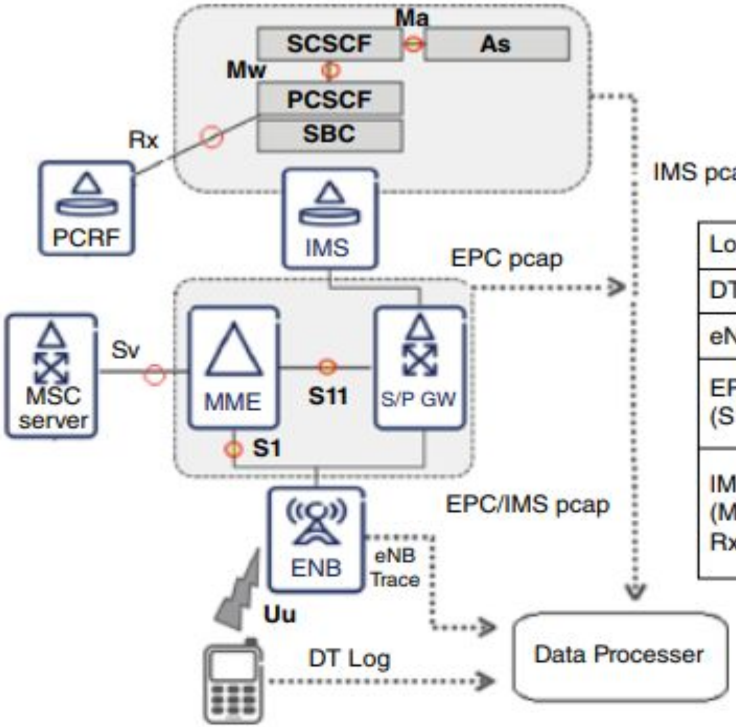
Optimization :

- For optimization, data capturing of VoLTE is necessary and involves :
- UU/X2 : filter control signalling
- S1-MME : filter traffic with associated IP address of eNB and MME.

Optimization :

- S1-U : filter IP address range for IMS APN + P-CSCF IP address
- Mw : filter control signaling between MSC server and ATCF
- ISC : for third party registration
- Sv : access transfer from PS to CS
- Rx : interface between P-CSCF and PCRF
- Gm : filter IP address of P-CSCF
- Mb : filter IP address range for IMS APN

Data capturing :



Log/Trace	Protocols
DT LOG	RRC
eNB Trace	RRC, X2AP, S1AP
EPC Trace (S1, S11)	S1AP,GTPv2, RADIUS, DIAMETER
IMS Trace (Mw, Ma,ISC, Rx etc.)	SIP, RADIUS, DIAMETER

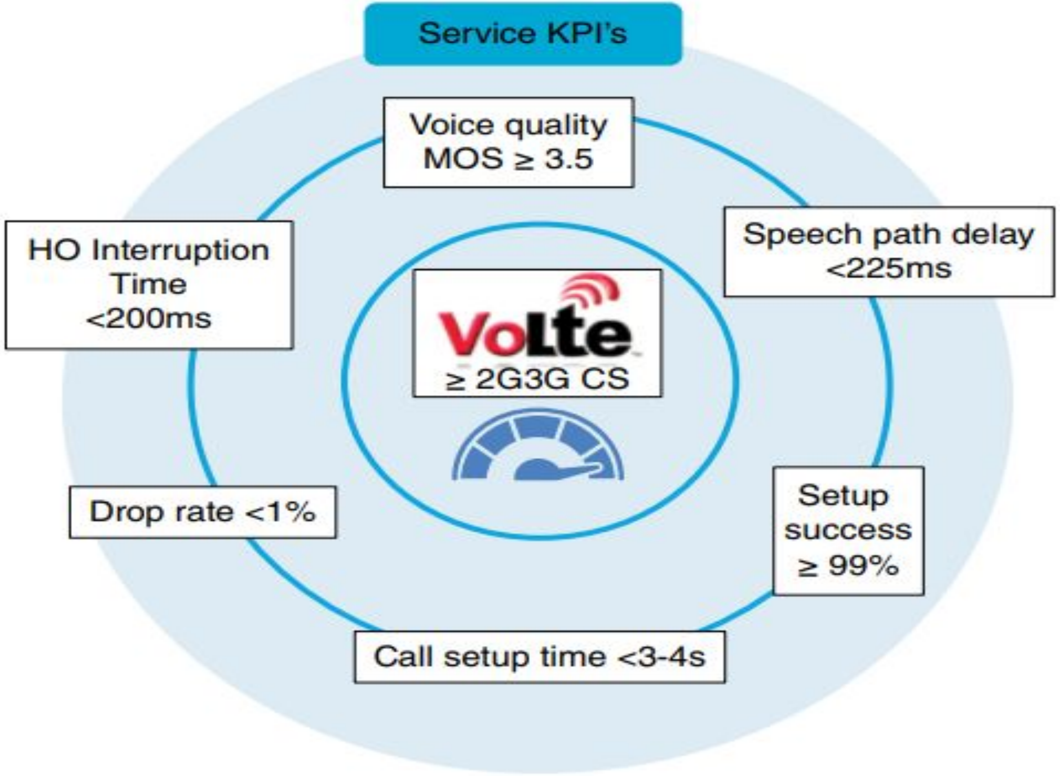
In order to analysis the voice performance as well as delay, interruption time and RTP loss, all SIP related signaling and media to and from the ATGW should be collected.

Data capturing :

For this, it needed

- RF statistics
- Logging RRC messages
- Logging IMS SIP message
- Logging RTP statistics each second

VoLTE service KPI’s :



- Cell Availability
- Connection Setup Time
- Connection Success Rate
- Attach Failure Rate
- RAB Setup Success Rate
- SIP Registration Success Rate
- Attach Setup Time
- SIP Registration Setup Time
- User Throughput
- Latency/Delay
- Handover Success Rate
- IRAT Handoff
- Packet Loss/Errors
- Jitter
- RAB Abnormal Release Rate
- RF Coverage (SINR)
- ...

Optimization :

- The voice quality a user perceived is determined by several parameters.
- From VoLTE services QoS perspective , there are several main service type KPI's to achieve : call setup time, RTP latency, jitter, quality, mouth to ear latency and handover interruption time.

KPI values :

- Call setup time : 900 - 4000 msec
- RTP latency : 30- 50 msec
- Quality : very good MOS score 3-4
- Mouth to ear latency : 180-280 msec
- HO interruption time : 45- 55 msec

KPI :

- Accessibility
- Retainability

VoLTE wireless issues signature :

Access failures reasons	Drops reasons
Bearer deactivation on both UEs	RLF-eNB detected
480 temporarily unavailable	Bearer deactivation on both the UEs
500 Internal server error	Potential product issue, RAN misconfigurations
487 request terminated	SRVCC failure
503 service unavailable	Handover failure
Non RF issue	Simultaneous SRVCC issue
QC1 not setup on both UEs	RLF-UE detected
Radio link failure-eNB detected	RRC connection re-establishment rejects
Attach failure	RTP timeouts including one-way audio for xx seconds
408 request timeout on MT UE (Call redirected)	SRVCC in alerting phases with CC NAS UNSPECIFIED
Create bearer request timeout followed by timeout failure	SRVCC failure with 404 not found
Create bearer request during X2HO	Call release by MSC with cause = 41 right after SRVCC
	Call released by SGC after registration with cause 404
	MSC rejects SRVCC PS to CS request due to incorrect STN-SR

Optimization

RB utilization :

- The utilization number of voice RB's is determined by scheduling cycle, voice quality and MAC padding overhead.
- The allocation method of voice RB's will impact VoLTE capacity and coverage which is needed to be optimized.
- Two RB's at QPSK are primarily used for UL RTP packets similar as downlink.

BLER issue :

- High BLER setting for VoLTE may result in higher packet loss and consequently poor signal quality.
- When UL BLER becomes 20%, VoLTE voice quality starts degradation.
- When UL BLER becomes above 50%, VoLTE call dropped.

Quality due to handover :

- The main handover failure root cause is RACH fail due to bad RF conditions or parameters.
- Before HO, SINR from source eNB is lower about -10db to 0 db.
- Decreasing the handover areas is the main task of mobility optimization.

eSRVCC handover issues :

- Call drops during SRVCC is the largest contributor to overall VoLTE call drop rate in live network.
- For SRVCC HO optimization, it is necessary to reduce the number of SRVCC's by providing adequate LTE coverage and not setting triggering threshold for SRVCC too early or too late.

Packet loss :

- RTP packet loss is normally due to poor RF, high loading of the cell, insufficient UL grant, handover and network packet drop etc.
- RTP packet loss observation can be done by RTP SN (sequence number) order.
- It is needed to be analyze if the packet loss is due to air interface , to check DL packets serial number is continuous or not.

Packet loss reasons :

- Packet loss due to poor RF
- Packet loss due to massive users
- Packet loss due to insufficient UL grant
- Packet loss due to handover
- Packet loss due to network issue

Call setup issues

Call setup issues :

- Call setup success ratio means the probability of being able to successfully invite a VoLTE session.
- It is one of the most important KPI for VoLTE service.

Call setup issues :

Call setup failure	Call setup time more than 6 sec
• Network not activate dedicated bearer after receive SIP 183 session progress • Network release dedicated bearer • MT is registering on IMS • No Paging to MT, paging issue, MT on GSM • RF issue • HO failure, aSRVCC failure • SIP message issues • IMS error (server_internal_error (500)) • redirection to GSM • RACH problem	• MT CSFB • frequent TFT updates/ modifications • Paging late • Network response for IMS msgis too late • Bad RF • RLF • activate dedicated bearer late • RACH problem

Missed pages :

- In mobility and dense urban environments, the usual case is that the MT UE frequently misses the first and second page due to idle cell reselection after the previous call ended , and finally the MT UE responds to the third page.
- If there is no response from the MT UE within 12 secs of the SIP invite message , the IMS network will send a SIP 480 (unreachable) to the MO UE.

IMS issues :

- VoLTE call setup failure due to server error.
- SIP message issue
- No response to SIP
- MO UE initiates to call when MT UE is doing registration on IMS.

Dedicated bearer setup issues :

- These issues usually happened after UE received SIP 183 session progress message, but network did not activate the dedicated bearer.
- Sometimes dedicated bearer released by network happened during handover process.
- When handover was triggered , the dedicated bearer that had established in source cell is released in the target cell.

CSFB call issues :

- In case, where one UE is VoLTE call, another one is CSFB call , this will lead to longer setup time (about 20 secs)

aSRVCC failure :

- If the feature of aSRVCC is not enable, which means SRVCC handover happened after SIP 180 ringing has been received by originating UE while terminating UE has not yet answered with SIP 200 OK.

RF issues :

- VoLTE call setup failure due to RACH problem.
- Long VoLTE call setup time due to RLF during call setup.
- MO UE tries to make VoLTE call , but RF is very poor.
- Call failure due to high BLER.

Frequent TFT updates :

- In dedicated bearer establishment procedure during call setup, resource allocation request is needed to create a new traffic flow template for dedicated bearer.

Encryption issue :

- When E-RAB establishment failure during call setup is analyzed, it is found some failures were due to because integrity and encryption protection algorithms are not supported.

Call drop

Call drop :

- The criteria for successful call completion is that the call is established successfully and in the end BYE and 200 OK (BYE) has been executed by two sides.
- A call drop indicates an abnormal release of the established call, which can be triggered by nodes in LTE, EPC, MSS and IMS with a variety of reasons.

Call drop :

- It usually acts as RRC abnormal release , dedicated bearer abnormal release , UE entered idle mode during the call or UE or IMS core transmits SIP cancel message.

Call drop :

- Call drop due to QCI-1 profile not defined
- Handover failure due to S1 path switch issue
- Handover failure (cell range issue)
- Call drop due to uplink issue

RTP-RTCP timeout :

- It is declared by telephony application server.
- UE re-establishment of RRC connection along with QCI-1 and QCI-5 was done successfully.

IMS session drop :

- SIP errors usually lead to session drop.
- Abnormal SIP BYE message will lead to UE session release.

Packet aggregation level :

- It is an important indicator of VoLTE performance.
- In downlink, packet aggregation depends on load and configured voice target delay.
- For 80 ms packet aggregation is mostly not needed and used quite rare.

VOIP OPTIMIZATION

VOIP Padding :

- A high percentage of MAC padding of VoLTE packet will degrade the system capacity, if it needs to pay more attention to uplink.
- It needs to be optimized, when MAC padding is higher than 50%.
- If grants is less than 20ms or more than 30 ms , UL transmissions will have more than 75% padding.

VOIP Padding :

- Some amounts of padding are expected due to RTP payload size and TB size granularity.
- For grants arriving less than 20 ms , a VOIP payload has not been generated and the grant must then be used with close to 100 % padding.

VOIP related parameters :

- LogicalchannelSR-mask
- SR-prohibit timer
- Short SR period
- CQI mask
- Explicit congestion notification
- Handover parameters (A3 parameters)
- RLC and HARQ parameters

Video- related optimization :

- Video telephony is the most challenging application because it needs stringent end to end delay requirements and multi-media synchronization.
- It is recommended to not use delay based schedular for video phone traffic since in case of poor radio conditions it effects users.

Handover parameters for VoLTE :

Parameter	Recommended	Comments
Triggerquantity	RSRP & RSRQ	Both RSRP and RSRQ triggering can allow for faster handover especially in loaded cells
reportQuantity	Both	
Hysteresis	4 (2 dB)	Small values can cause cases of going out of triggering conditions earlier and TTT reset
a3-offset	6 (3 dB)	Low value would cause more handovers and possible ping pongs impacting VoLTE interruption
timeToTrigger	240 ms	Higher value delays handover
ReportAmount	1	It stopped after only one measurement, otherwise causing delayed handover especially if there is no reply to the first MRM
filterCoefficient	fc8-11	If this value is set too low, measurement reports could be triggered by rapid, temporary, short term fluctuations in RSRP
ReportInterval	240 ms	Currently does not apply since only ReportAmount = 1

Video- related optimization :

- In video telephony service, the VOIP and video streams are sent synchronously but carried on separate bearers with different QCI's and different delay budgets.
- If the video stream fails to get established , the call can continue as VOIP only be terminated.
- Audio and video synchronization can be achieved in the UE by using timing information carried in RTCP packets.

Video bitrate and frame rate :

- A too low bit rate will produce too low frame rate for the particular screen/resolution.
- Video bitrate on IP level can be calculated from the IP traces.
- Video frame rate is one of the important KPI for end user perception of video.
- Frame rate can be calculated based on IP traces using frame marker in RTP header.

Video MoS and audio/video sync :

- Perceived video quality is content dependent.
- The A/V sync is the offset in delay between audio and video is of crucial importance for the quality perceived by the end user.

UE battery consumption optimization for VoLTE

Optimization :

- UE battery consumption depends on background activity, device hardware and network settings.
- Power savings is possible in idle as well as connected mode.
- For higher battery savings , a combination of long and short DRx can be used.

Connected mode DRx parameter :

- In order to fulfill low battery consumption requirement for VoLTE , short on duration & DRX inactivity timer is necessary.
- Operator can configure upto 5 DRx profiles.

DRx optimization :

- State estimation
- DRx optimization and parameters
- KPI impacts with DRx

State estimation :

- If there is reception of consecutive non-speech frames , the bearer is transitioned to speech inactive state.
- The decision of speech inactive to speech active transition does not only rely on size of RLC SDU’s.

DRx optimization & parameters :

- Selection of DRx profile is the main task of DRx optimization.
- It depends on UE capability, allocated bearers, balancing UE power consumption and signalling load.
- In VoLTE, QCI 1 and non QCI 1 are setting different DRx profiles.

DRx optimization & parameters :

Explanation of DRx operation during each phase of VoLTE call procedure :

- During short time (about 300 ms) there is no DRx.
- During time (500 ms) UE uses a default DRx , but UE is active so DRx sleep ratio is 0%.
- During time (10 sec) UE uses a default DRx , the UE is inactive -> DRx sleep ratio is 97%.
- During time (about 4 mins) of VoLTE conversation , DRx sleep ratio is 80 to 85%.

KPI impacts with DRx :

- DRx is an ideal feature to save the UE battery under low network load and good radio quality.
- Call drop rate increases when DRx is activated if no special handling is done in the eNB.

Comparison with VoLTE and OTT

Comparison :

- OTT refers to services provided independently over the mobile operators.
- Many OTT applications that support VOIP include rich communication features like instant message, file sharing, video calls etc.

OTT VOIP user experience :

- OTT VOIP can provide better speech quality.
- VoLTE voice quality is not affected by non-GBR traffic because of QoS configuration, for OTT VOIP calls when high non-GBR load causes upto 65%, the calls started drop.

OTT VOIP codec :

- Silk codec is used by some OTT voice service.
- Silk codec has four operating modes.

Signaling load of OTT VOIP :

- OTT communication services are particularly challenging, because always on reachability requires a persistent connection to the network.
- OTT VOIP applications are a common reason for frequent data transactions.
- Frequent RRC state changes increases the signaling load in the whole mobile networks.

PRACH Optimization

Use of random access in LTE :

- Initial access or tracking area updates when establishing a radio link.
- To re-establish a radio link after a radio link failure.
- During the handover process to establish UL synchronization to the target cell.
- For positioning purposes, when timing advance is needed for UE positioning.

RACH optimization :

- It can be done by OMC statistics, like preamble allocation, random preamble detection, PUSCH loading information and so on.
- If the eNB receives a small number of random preambles and current PUSCH is high loading , it needs to reduce the PRACH resources in each radio frame.

PRACH configuration index :

- PRACH transmission is composed of a preamble sequence and a preceding cyclic prefix with five different formats.
- PRACH configuration index parameter specifies the index, which informs UE of which frame number and sub frame number within the frame has PRACH resources.

RACH root sequence :

- The logical RSI is used to create a random preamble.
- In case of RSI collision, it is recommended to change RSI on either of the site.
- RACH root sequence planning is dependent on PRACH cyclic shift.

PRACH cyclic shift optimization :

- The cell range i.e the requirement for guard period defines the length of cyclic shift.
- Cyclic shift dimensioning is very important in RACH design.
- eNB should configure Ncs independently in each cell.

Random access issues :

- Three kinds of UE may do random access.
- In situation of low coverage or high interference , it will cause bad link quality, so coverage and interference optimization needed.
- Contention failure.

RACH message optimization :

- The message could be any as RRC connection request, RRC connection reconfiguration complete or an RRC connection re-establishment depends on the cause which initiated the RACH procedure.
- In the worse RF condition, the message's coverage may be improved with RLC segmentation for fragmenting this handover complete message.

RACH message optimization :

- The message could be any as RRC connection request, RRC connection reconfiguration complete or an RRC connection re-establishment depends on the cause which initiated the RACH procedure.
- In the worse RF condition, the message’s coverage may be improved with RLC segmentation for fragmenting this handover complete message.

Accessibility Optimization

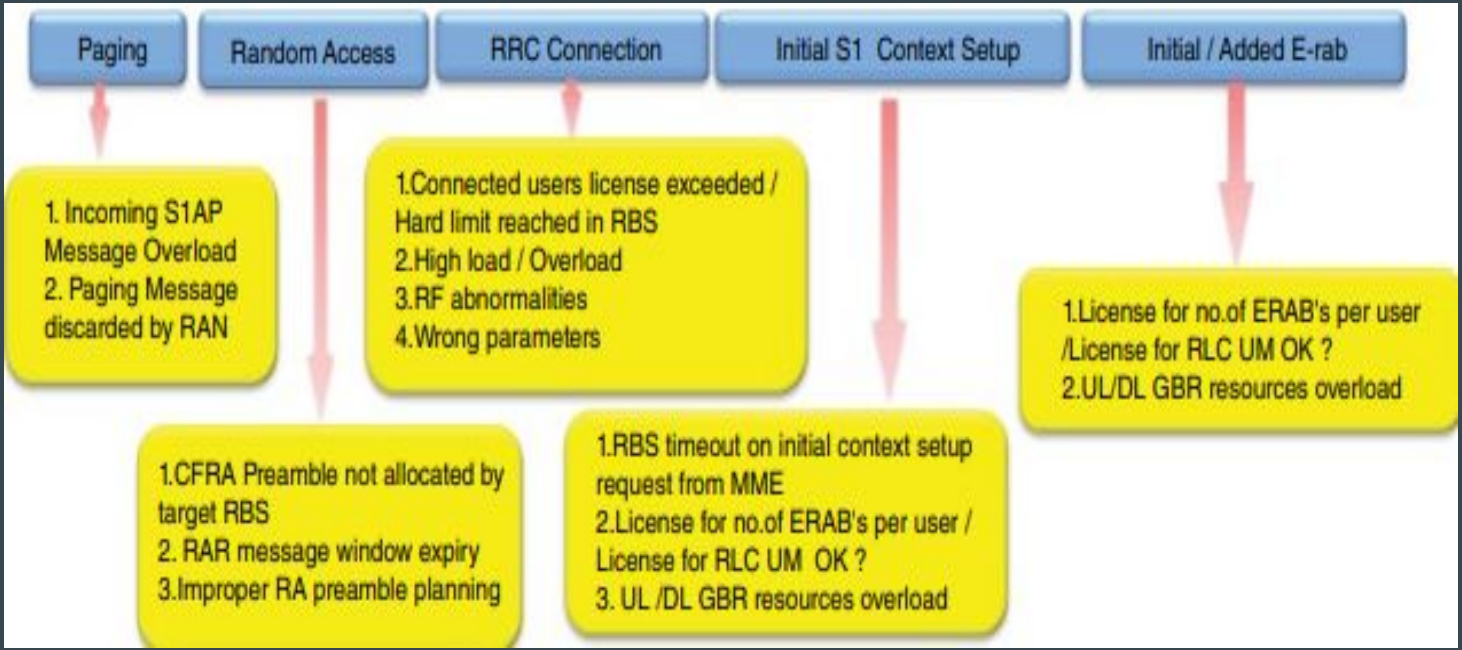
Reasons for poor accessibility :

- Poor coverage
- High load
- Hardware issue
- High UL interference
- PCI conflict
- RACH root sequence index plan

Accessibility :

- UE initial access from RRC_idle state.
- RRC connection re-establishment procedure and handover.
- DL data arrival during RRC connected state requiring random access procedure, when UL synchronization status is “non-synchronized”.
- UL data arrival during RRC connected state requiring random access procedure, when UL synchronization status is “non-synchronized”.
- There are no PUCCH resources for SR available.

Accessibility Analysis tree :



Accessibility analysis tree :

Accessibility measurements can be based on drive tests or signaling statistics. Reasons for poor accessibility :

- Poor coverage
- UE camping in the wrong cell
- High UL interference
- The preamble collision
- Transport related issue
- Admission reject

Call & data session setup optimization :

- Call setup has 5 phases.
- Success of phases 1-2 must be monitored from PRACH & RRC signalling counters.
- Success of phases 3-5 monitored from S1AP and E-RAB counters.
- Bearer establishment has three steps.

PCI Optimization

Overview :

- PCI is used to indicate the physical layer identity of the cell.
- There are total of 504 ID's (0-503).
- PCI's are grouped in 168 PCI groups.
- After UE powered on, first thing is PCI acquirement.
- PCI mod 3 and PCI mod 6 values are applicable for single transmit antenna configuration.

PCI optimization methodology :

- Purpose of PCI optimization is to ensure that neighboring cells should have different primary sequences allocated.
- Good PCI assignment reduces call drops by enabling UE to clearly distinguish one cell from another.
- Cell ID planning means realizing P-SCH and S-SCH planning.

PCI check & optimization methodology :

- Check that no neighboring cell has the same PCI value.
- Check that no neighboring cell of any neighbor cell has an identical PCI value.
- Check if another cell within a specific distance has an identical PCI value.
- Check if another cell within the signal strength threshold has an identical PCI value.

PCI group optimization :

- In a live network , from all the available PCI's, the groups are used for PCI planning in the LTE network.
- This allows for 126 PCI's per group.
- 120 per group available for the planning of the outdoor macro LTE system.
- 6 per group reserved for growth within the group.

PCI reuse distance :

- If there are more than 504 cells, the PCI's have to be re-used.
- The co-PCI cells should be as far apart as possible.
- The distance between the same PCI code being used at one cell and then reused at the second cell should be as large as possible.
- In the context of a rural network, aim for a re-use distance should be bigger than 2*maximum cell radius.

PCI collision and confusion :

- PCI collision can also happen when a cell and its defined neighbor use the same PCI code.
- PCI confusion is a case when a serving cell has got two neighbors with the same PCI.

PCI optimization :

- The aim of PCI optimization is removing the collision and confusion of PCI based on neighbor relation and minimizing RS DL/UL conflict.
- Periodically PCI monitoring and optimization are needed, which will provide a healthy network.

Tracking areas Optimization

TA optimization :

- The network is organized into groups of cells called tracking areas.
- When a UE registers itself in the network, the UE is responsible for registering itself within a specific TA, the core network stores information about TA where registration is performed.
- SIB 1 broadcasts the TA to which a cell belongs.

Cases of TA update :

- UE entered a new TA that is not in the list of TAIs that the UE registered.
- The periodic TA update timer has expired.
- UE was in UTRAN connected state when it re-selects to LTE.
- The TIN indicates P-TMSI when the UE re-selects to LTE.
- RRC connection released
- UE core network capability or UE specific DRx parameters has been changed.

TA design :

The key for tracking area design should consider these factors :

- The number of eNB's in a TA.
- Low UE power consumption
- Low paging delay
- Low network load

TA optimization :

- The main target of TA optimization is to minimize paging and TA updates.
- It is necessary to achieve balance between number of TA updates and paging load.
- There are two methods of initial TA planning.

TA update procedure :

- The new MME contacts the old one via GTP-C context request message.
- With one of the authentication vectors the new MME can start authentication.

Tracking areas Optimization

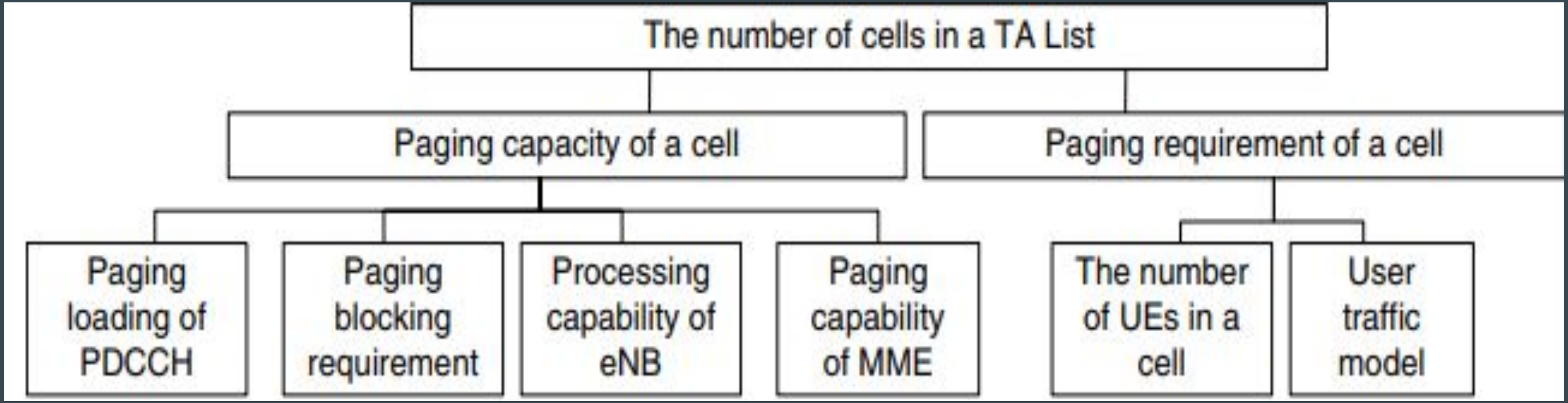
TA optimization and TAU failure :

- The paging channel is able to support 1600 pages per second.
- The main benefit associated with large TAs is a reduced requirement for TA updates resulting from mobility.
- TA updates generate signaling and also increase the probability of a UE missing a paging message.
- Method of avoiding frequent TA update is that tracking areas should not run close to and parallel to major roads.

TA list optimization :

- In the area of TA management, the LTE systems admit more flexible configuration.
- The MME will automatically create TA list based on UE mobility, and sends to the UE a TA list containing the current TA and one or several neighbor TAs, also can update the TA list to avoid ping pong events at TA borders.

TA list optimization :



TA list optimization :

TA list design can be divided into two kinds of methods :

- The TA area is determined by the traditional way, the no. of TA's in the TA list is necessary less, for 1 to 3 and so.
- TA list optimization is in accordance with a single UE.

TAU reject analysis and optimization :

- Network failure
- Implicitly detached
- No EPS bearer context activated

UL signal Optimization

UL signal optimization :

Two types of reference signals are defined in LTE uplink :

- DMRS
- SRS

UL reference signal optimization :

- LTE PUSCH and PUCCH carry DMRS.
- On PUSCH , each slot contains one DMRS SC-FDMA symbol.
- On PUCCH, correct detection of carried control data is equally important for all terminals.

UL sounding signal optimization :

- UL FSS is important and applied in LTE that allocate the spectrum for PUSCH transmissions.
- SRS can be transmitted even when no PUSCH is transmitted.

Hetnet Optimization

Hetnet optimization :

- Hetnet is a general term for a network that consists of different types of cells working together as one seamless network.
- Hetnet is defined as placing lower power nodes in the macro coverage area, the distribution of nodes can be depending on several factors either covering coverage parts or in hotspots are for densification of capacity.

UE geolocation and identification of traffic hotspots:

- With assessment of geo - location , it need to find the location of traffic hotspots from the field geo-located subscriber records.
- Small cell coverage size is in the range of 50-100 mts.
- Distance between small cells depends essentially on UL service requested, or on the capacity required in term of users/km2.

Wave propagation characteristics :

- The received signal at any location depends on the proximity of the mobile to buildings that may partially block the signal path between the transmitter and receiver.

New features in hetnets :

The features include :

- Small cell discovery
- CoMP
- UE speed estimation
- Improved RLF discovery
- Handover command protection by ABS (almost blank subframe)

Combined cell optimization :

- The combined cell configures multiple sectors (RRU's) to serve same cell.
- It aims to extend the indoor or outdoor coverage of a single cell.
- The benefits of the feature include reducing coverage holes by allowing multiple coverage areas within the same cell.

Signaling based Optimization

Signaling based optimization :

- Worst cells needs to be monitored and treated continuously.
- Signaling messages to be traced over the specific interfaces for specific UE's.
- LTE signaling monitor captures mass data from customer live network through the interface Uu, S11, S1-MME, S1-U, S6a, SGs and make collections of the responding KPI's.

S1-AP Signaling :

- There are three important interfaces to be analyzed for network optimization, Uu, X2 and S1.
- S1AP signalings are : SAE bearer management, UE context management, S1 interface management, mobility functions, NAS signaling transport, paging.

NAS Signaling :

- NAS consists of the session management and mobility management layers.
- Functions performed by NAS messages include mobility management for idle UE's , UE authentication, EPS bearer management, configuration and control of security, paging initiation of idle UE's.

UE Signaling management :

- The length of idle timer will impact on terminal behavior and network signalling load.
- 80% of signaling is coming from service requests.